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Postal address

3GPP support office address
650 Route des Lucioles - Sophia Antipolis
Valbonne - FRANCE
Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Internet
http://www.3gpp.org

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Contents

Foreword

This Technical Specification has been produced by the 3GPP.

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of this TS, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version 3.y.z

where:

x the first digit:

1 presented to TSG for information;

2 presented to TSG for approval;

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y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the specification.

1 Scope

This Technical Specification specifies the stage 2 description of the Out-of-Band Transcoder Control for speech services.

Cellular networks depend heavily on codecs to provide their services. Codecs are necessary to compress speech in order to utilise efficiently the expensive bandwidth resources both in the radio interface and in the transmission networks.

Unnecessary transcoding of speech significantly degrades quality and, therefore, cellular systems try to avoid it for mobile-to-mobile calls when both UEs and the network support a common codec type.

Digital cellular systems support an increasing number of codec types. As a result, in order to allocate transcoders for a call inside the network, and to select the appropriate codec type inside the UEs, signalling procedures are defined to convey the codec type selected for a call to all the affected nodes (UEs and (potential) transcoding points inside the network). Also, codec negotiation capabilities are being defined to enable the selection of a codec type supported in all the affected nodes, i.e. to resolve codec mismatch situations. This codec negotiation maximises the chances of operating in compressed mode end-to-end for mobile-to-mobile calls.

Although the main reason for avoiding transcoding in mobile-to-mobile calls has been speech quality, the transmission of compressed information in the CN and CN-CN interface of the cellular network also offers the possibility of bandwidth savings.

To allow transport of information in a compressed way in transmission networks, these networks make use of thetransport -independent call control protocol as specified in [8] that provides means for signalling codec information, negotiation and selection of codecs end-to-end.

2 References

The following documents contain provisions that, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[1] 3G TS 23.107: "QoS Concept and Architecture"

[2] 3G TS 24.008: "Mobile radio interface layer 3 specification Core Network Protocols –Stage 3"

[3] 3G TS 25.413: "UTRAN Iu Interface RANAP Signalling"

[4] 3G TS 25.415: "UTRAN Iu Interface User Plane Protocols"

[5] 3G TS 26.103: "Speech codec list for GSM and UMTS"

[6] Q.1902.x: "Bearer Independent Call Control, CS2"

[7] Q.765.5:" Application Transport Mechanism for Bearer Independent Call Control"

[8] 3G TS 23.205, Bearer-independent CS Core Network.

[9] 3G TS 33.106, Lawful Interception Requirements

[10] 3G TS 28.062, Inband Tandem Free Operation (TFO) of Speech Codecs.

[11] 3G TS 23.009, Handover Procedures.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of this specification the following definition apply:

Codec: A codec is a device to encode information from its original representation into an encoded form and to decode encoded information into its original representation.

Tandem Free Operation: Tandem Free Operation is the configuration of a connection with two transcoders that support TFO protocol and whose external coding schemes are compatible, thus enabling compressed speech to pass between them. When the TFO protocol is not supported by both transcoders or the coding schemes are not compatible then normal "Tandem" operation occurs and PCM encoded speech is passed between them.

Transcoder: A transcoder is a device to change the encoding of information from one particular encoding scheme to a different one. Most commonly to/from a compressed speech algorithm from/to PCM.

Transcoder Free Operation: Configuration of a speech or multimedia call for which no transcoder device is physically present in the communication path and hence no control or conversion or other functions can be associated with it.

Out of Band Transcoder Control:

Capability of a system to negotiate the types of codecs and codec modes on a call per call basis through out-of-band signalling. Out-of-Band Transcoder Control is required to establish Transcoder Free Operation.

Default PCM Codec: This is the network default codec for speech in PCM domain, for example ITU G.711.

Transcoding free link (TrFL): A transcoding free link is a bearer link, where compressed voice is being carried between bearer endpoints. Within the UMTS network, the compressed voice is transmitted in Iu User Plane format.

Tandem free link (TFOL): A tandem free link is a bearer link between transcoders that are operating in Tandem Free Operation mode, i.e. bypassing the transcoding functions. The involved transcoders can be a UMTS transcoder or a GSM TRAU with TFO functionality.

Transcoder free operation (TrFO):

This term is applicable to calls that have no transcoders involved in the connection between the source codecs. For mobile to mobile calls this is UE to UE, although the connection could be UE to another type of terminal. TrFO operation is considered a concatenation of TrFLs between RNCs.
In case of mobile to fixed network calls the term “Transcoder free operation” is applicable for the TrFLs carrying compressed speech. The TrFO usually ends at the Gateway to the PSTN where the speech is transcoded e.g. to G.711.

Tandem free and Transcoding free operation (TaTrFO): Tandem free and transcoding free link operation is the concatenation of "transcoding free links" and "tandem free links".

3.2 Abbreviations

Abbreviations used in this specification are listed in GSM 01.04.

For the purposes of this specification the following abbreviations apply:

APM Application Transport Mechanism

BC Bearer Control

BICC Bearer Independent Call Control

CC Call Control

CCD Conference Call Device

CFNRy Call Forward on No Reply

CFNRc Call Forward Not Reachable

IN Intelligent Network

OoBTC Out-of-Band Transcoder Control

QoS Quality of Service

RAB Radio Access Bearer

TFO Tandem Free Operation

TICC Transport Independent Call Control

TrFO Transcoder Free Operation

UP User Plane

4 Out-of-Band Transcoder control functionality

4.1 OoBTC Requirements

The OoBTC mechanism shall support the following:

- The capability to negotiate the preferred codec type to be used between two end nodes and to avoid the use of transcoders in the network at call set-up.

  The originating UE indicates the list of its supported codec types for codec negotiation. This list shall be conveyed to the terminating MSC. The terminating UE indicates its list of supported codec types to the terminating MSC.

  Where no compatible codec type can be selected between the UEs then the default PCM coding shall be selected. The originating MSC shall insert a transcoder in the path from the originating UE. Codec selection for the terminating UE is then performed within the terminating MSC, independently of the originating MSC.

  Note: For a codec type supporting various modes, the described functionality shall also be applicable to negotiate the set of codec modes common to originating and terminating UEs. Other negotiations such as Initialisation and Rate control are performed at a later point in time by the Iu UP protocol.

- The capability to control the presence of transcoders in the network after call set-up.

  Where a change to the call state of a transcoder free connection occurs, such that compressed speech cannot be maintained, it shall be possible to insert a transcoder or pair of transcoders where needed in the path. If this results in change to the encoding of the speech in other nodes then it shall be possible to inform the end points of this segment that the speech coding is changed. Such examples where this could occur are:

- SS interruptions (e.g. A to B call connection becomes to multiparty call connection.)
-
- Handover to an incompatible partner.
- Synchronisation loss

Where a change in call state as described above is temporary then it shall be possible to return to a transcoder free connection by removing the inserted transcoders and informing the endpoints that the connection has resumed to compressed speech encoding.

- The codec types comprise codecs for speech in the first phase. The transcoder control should have enough expandability to support future enhancements of codec types.
- The transcoder control procedure shall not cause a perceivable time lag in the cases of establishing transcoder free connection and reverting to normal (double transcoded) call connection in the cases described above for control of the presence of transcoders.
- The capability to insert transcoder (in cases where a TrFO connection is not possible) at the most appropriate location, i.e. to save bandwidth it should be located at the CN edge between an ATM or IP transport network and a STM network.
  - When a transport network cannot maintain compressed voice then reversion to the default PCM coding shall occur. A transcoder shall be inserted at that point and OoBTC procedures terminated. TrFO link is then possible between that point and the preceding nodes.
  - When a Non-TrFO call reaches the UMTS CN then OoBTC procedures are initiated from that point and after codec negotiation has been performed, if compressed voice can be supported through the CN then a transcoder is inserted at the edge of the CN.
- The OoBTC signalling procedures shall be supported by the call control protocol on the Nc interface, for example codec negotiation, codec modification, codec list modification, codec renegotiation, and codec list re-negotiation. BICC CS2 [6] supports such a mechanism, through the APM procedures defined by [7].

4.2 Relationship between OoBTC and In-band TFO

OoBTC is used before call set-up to attempt to establish an UE-UE transcoder free connection. If successful the result is a saving of transcoding equipment in the path and provides a cost efficient transmission.
The In-band TFO protocol is activated after call set-up only if transcoders are inserted in the path. In case two transcoders in tandem (a pair of transcoders with PCM coded between them) are able to communicate to each other (both support TFO), then the inband TFO protocol allows the transcoders to compare coding schemes. If compatible codec types exist, the transcoders are able to overwrite the PCM with the pure compressed speech (effectively bypassing the transcoding functions). In-band TFO provides fast fallback mechanisms in case the TFO connection can not be maintained (insertion of CCD, DTMF, tones, etc). In-band TFO provides no direct saving of transmission costs.

If the OoBTC fails to establish the TrFO and transcoders are required, then in-band TFO may be used after call set-up. Inband TFO shall be the fallback mechanism when transcoders cannot be avoided, either at set-up or during the communication phase. In-band TFO shall be used for interworking with the 2G systems (e.g. GSM).

4.3 Lawful interception

The TrFO shall be maintained if the interception is made due to the lawful interception. Two decoders are needed to monitor the TrFO call.

Lawful interception shall not have any influence on the establishment or maintenance of the TrFO connection in order to avoid any audible effect in speech quality or noticeable effect in speech delay to the end users.

The existing requirements for lawful interception shall be considered, these are described in [9].

5 General Principles

5.1 Network Model

The codec negotiation mechanism (OoBTC) is designed to work in the general situation where more than two call control (CC) nodes need to participate in the codec negotiation. The codec negotiation mechanism works as follows:

- Originating CC node: sends its list of supported codec types and options, listed in order of preference.
- Transit CC nodes: if needed, analyse the received list of options, delete unsupported options from the list and forward the list. No modification is done to the preference levels of any of the listed codecs.
- Terminating CC node: analyse the received list of options with their associated priorities and selects the supported option with highest indicated priority.

Figure 5.1/1 illustrates the architecture for R00 for UMTS to UMTS TrFO connection. The transit network may exist for calls between PLMNs or between islands of mobile CNs separated by transit networks. This figure is a basic illustration, OoBTC shall apply to other access technologies where the OoBTC procedures are supported, i.e. not limited to this figure. The negotiation occurs at call set-up phase, and possibly later on in the call due to other changes such as handover or relocation. <u>However, as described in the next section, it shall be possible to modify the selected codec at any moment during the active phase of the call.</u>

Further detail of the Call & Bearer Separation for 3GPP is described in [8].

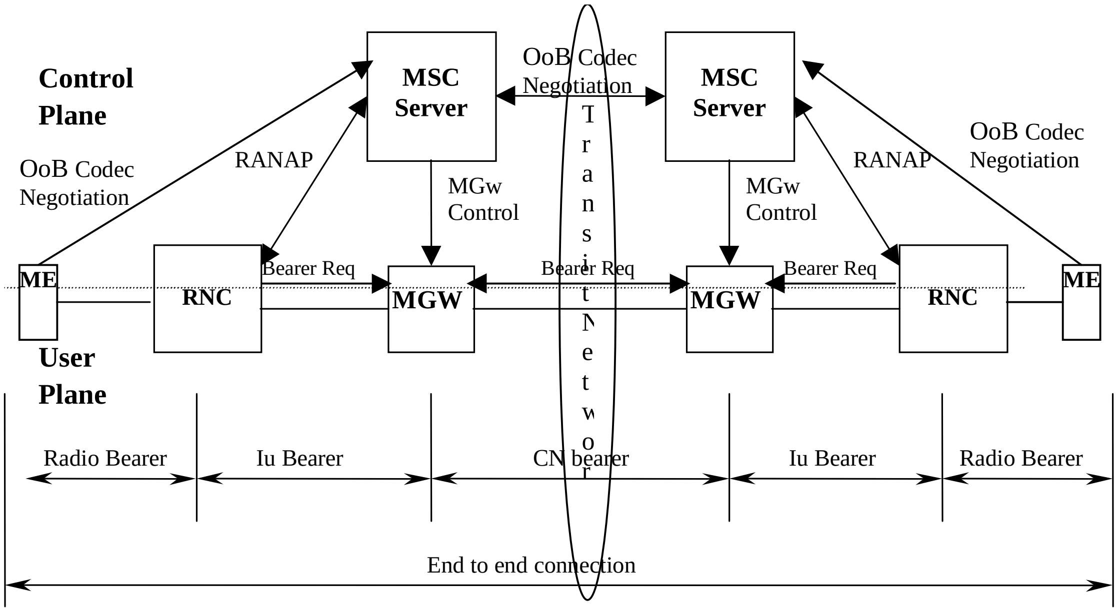


Figure 5.1/1. Basic Architecture for UMTS to UMTS TrFO Connection

The following sections describe successful call establishment scenarios using the codec negotiation mechanism.

5.2 Simple call set-up

The signalling flow for the simple call set-up case is illustrated in figure 5.2/1. Codec negotiation is done prior to the establishment of bearer connections, so that appropriate bearer resources are committed to the call. In the proposed sequence, the codec negotiation starts with the IAM message containing the list of supported codec types (in this example v, w, x, y, z), sent by the Originating MSC (O-MSC). Transit nodes may puncture out (i.e. delete) codec types from the list (in this example y). The terminating MSC (T-MSC) selects the codec type (here x) The selected codec is conveyed in an APM message, together with the remaining list of alternative, but currently not selected codec types (here v, x, z).

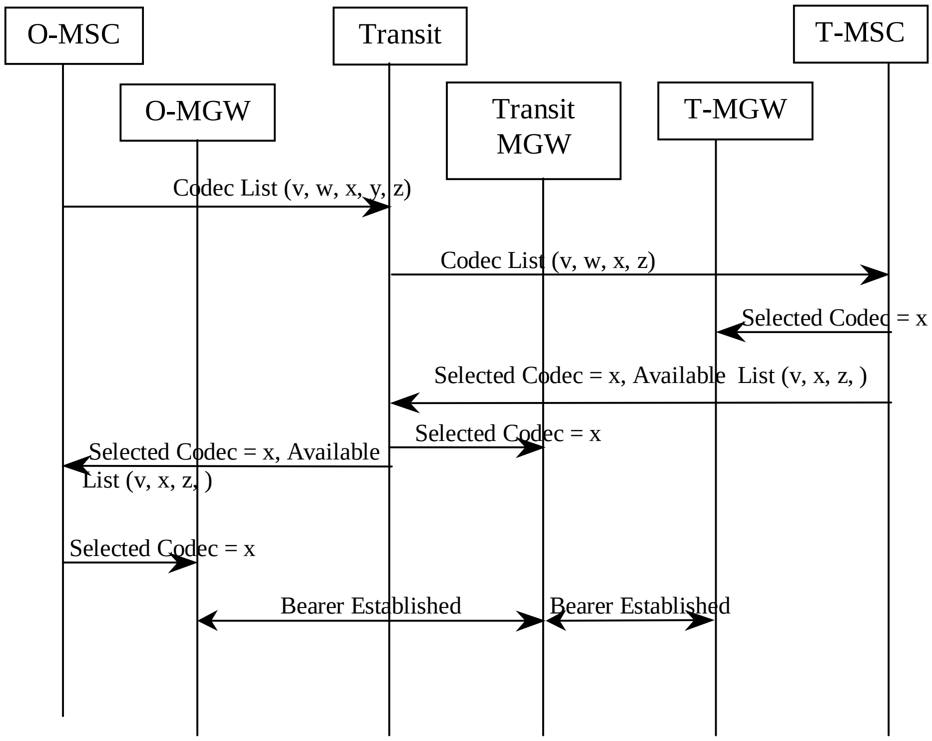


Figure 5.2/1. Basic Codec Negotiation Sequence

The codec list for BICC is specified according to [7], where each 3GPP codec entry is defined according to [5].

5.3 Media Gateway Control for Codec Handling

The general handling of MGW control procedures are detailed in [8]. Specific handling related to the control of the speech encoding is detailed in Figure. 5.3/1

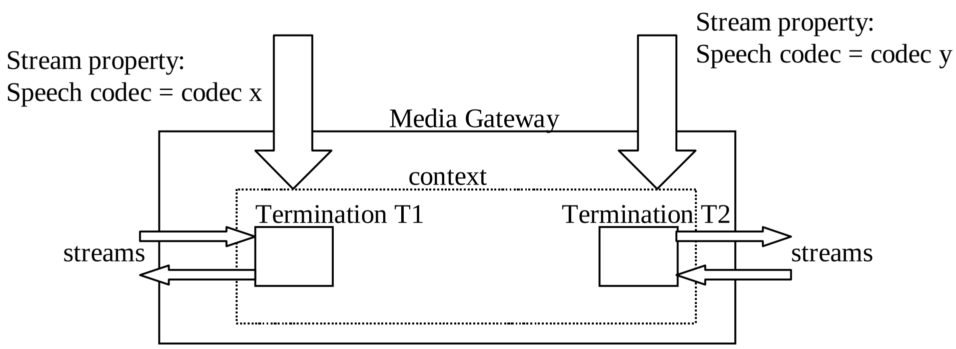


Figure 5.3/1. MGW control for speech codec

The handling of transcoding between one codec type (media stream property applied at one termination) and the another codec type (media stream property at other termination) is a function of the MGW. The media stream property for Audio Codec Type is defined in Annex C of the ITU-T MGW control protocol, H.248.

5.4 Iu UP Framing Protocol Handling for TrFO

5.4.1 Framing Protocol Initialisation

For TrFO calls the compressed speech is carried end to end (RNC to RNC or between RNC and other compressed voice terminal). In 3GPP Core Networks compressed voice shall be carried using the Iu User Plane Protocol. The specification for Iu interface is defined in [9]. For compressed voice only the support mode is used, thus for TrFO the Iu UP Initialisation procedure shall be supported by the CN, when a CN MGW is required to establish a connection with the Iu UP protocol.

The Iu UP Protocol is established through the CN in a forward direction, independently of the bearer establishment direction. The Notify message to indicate bearer establishment shall not be sent until the Iu UP has been initialised. The continuity message (COT) shall not be sent forward until the Notify message has been received from the MGW and also the COT from the previous server has been received. The sequences for mobile originated calls are shown in figures 5.4/1 and 5.4/2 for forward bearer and backward bearer establishment, respectively. The parameters in the Add Request messages in the Figures are described in further detail in chapter 5.4.5.

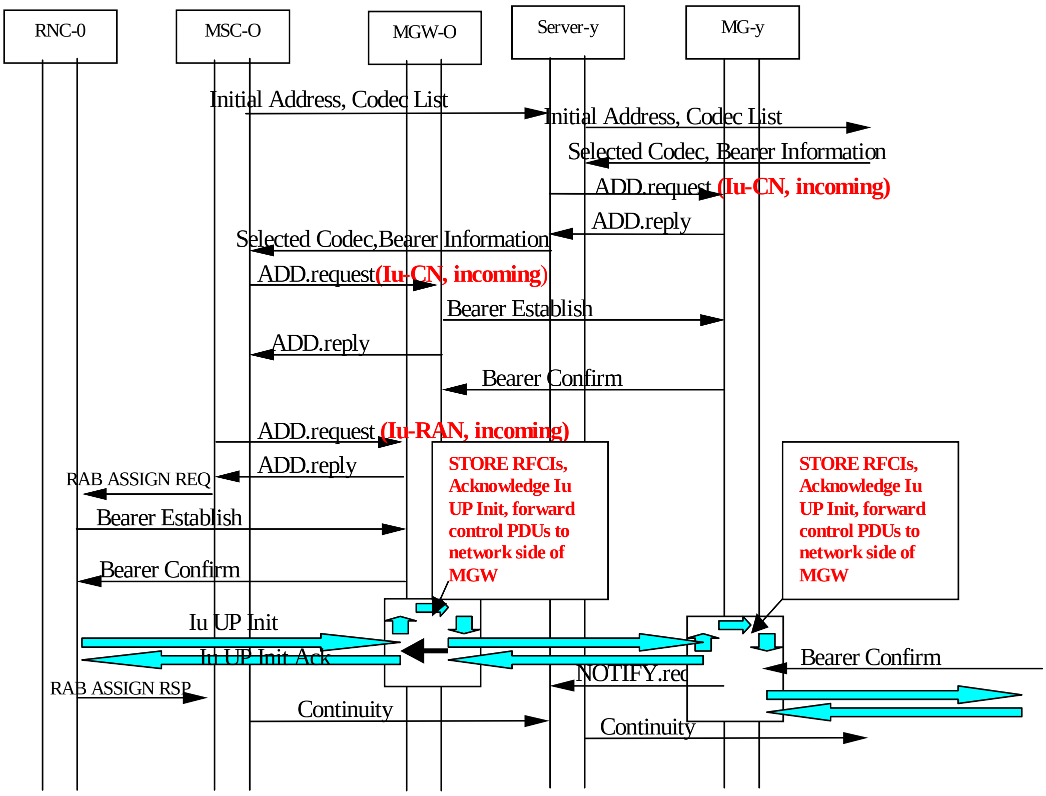


Figure 5.4.1/1: Iu UP Protocol Establishment, Mobile Originating call, forward bearer.

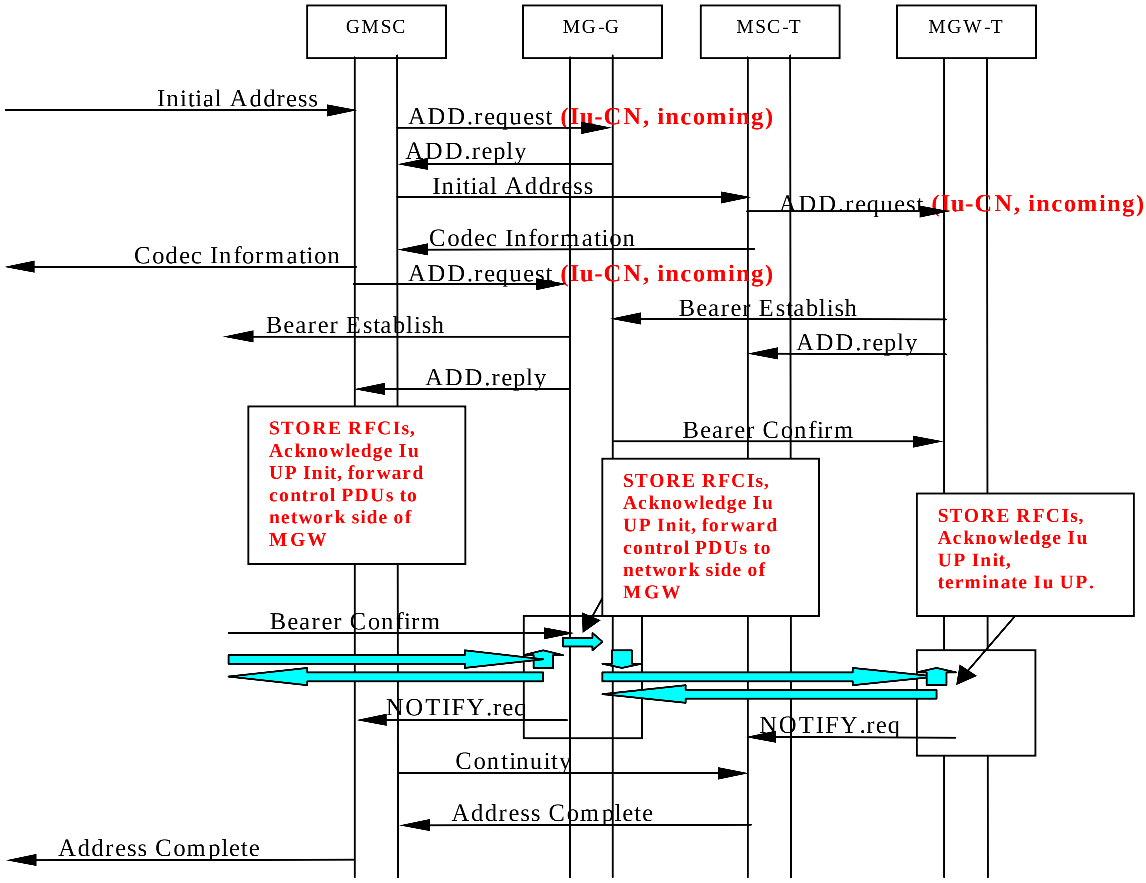


Figure 5.4.1/2: Iu UP Establishment, backward bearer.

The transport independent call control procedures in [8] shall support a continuity mechanism, as described above.

5.4.2 RFCI Storage

RAB Subflow Combination Identifiers are allocated to the SDU formats sent to the RNC by the MSC in the RAB Assignment. This allocation is then sent in the Iu UP Initialisation PDU by the RNC in the User Plane. For further details see [3] and [4].

During the TrFO call establishment each MGW linked into the call shall store the RFCIs received from Iu UP PDU Type 14.

After the out of band codec negotiation has been performed, if the originating side is a UTRAN, then on request from the MSC for a RAB Assignment, it shall initiate the Iu user plane. If the originating side is a network that does not support Iu UP then the Iu UP initialisation is initiated by the GMSC, as described in detail in Chapter 6.7. An Initialisation Protocol Data Unit (PDU) shall be sent to the first MGW in the call connection. Each initialisation leg is acknowledged per TrFO Link, i.e. per MGW-MGW interface. The subsequent initialisation is performed using the same RFCI set as received from the preceding node, independently of the Stream mode directions (i.e. if the terminations are not through connected).

This is shown figure 5.4.2/1.

Figure 5.4.2/1: RFCI Storage and subsequent initialisation in MGW

When the MGW terminations are through-connected and the RFCIs at both terminations are matching, then the MGW may revert to transparent mode; the RNCs shall not perform any subsequent Iu UP initialisations without explicit request by the serving MSCs.

All succeeding MGWs in the path shall behave in a similar way as described above.

5.4.3 RFCI Mismatch Resolution

At the terminating end of a TrFO connection with Iu UP initialised to the terminating MGW, the originating RFCI allocation is stored. The terminating RNC is then requested to perform a RAB Assignment towards the terminating MGW. This results in an Iu UP initialisation, where the allocation of the RFCI values is independent from the Originating RNC's allocation. These values may then be different to the originating RNC's set.

The terminating MGW shall acknowledge the Iu UP Initialisation and compare the RFCI values stored from the originating side. If the allocated index values do not match then the MGW shall initiate an Iu UP Initialisation PDU towards the terminating RNC with the RFCI allocation as defined by the preceding node (previously stored in the MGW). This is shown in figure 5.4.3/1

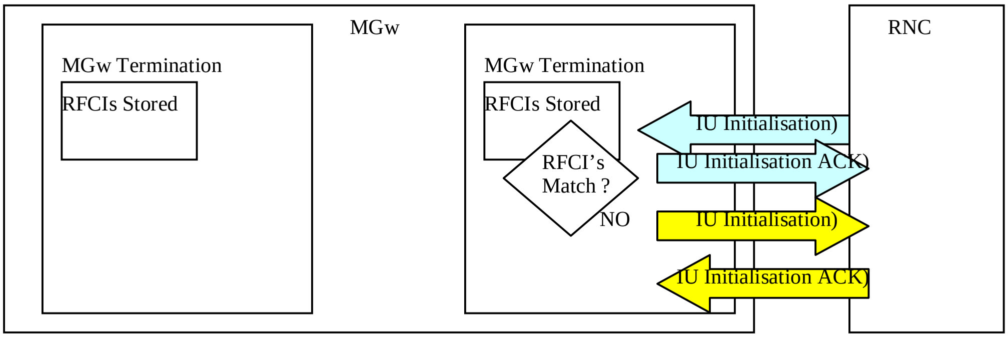


Figure 5.4.3/1:RFCI Mismatch Resolution

Further details of the TrFO call establishment are described in chapter 6.

This resolution handling is required also during RNC relocation; further details are described in chapter 6.

5.4.4 TrFO Break

The event and procedure when a TrFO connection must be interrupted at a certain point in the path, e.g. due to a supplementary service invocation or for handover/relocation, is termed "TrFO Break". A TrFO Break occurs at a MGW directed by the appropriate Sever. During this period the Iu User Plane protocol is terminated by this MGW, in general at both sides of the MGW. This means that it must respond to new Initialisation PDUs and Inband Rate Control PDUs. The MGW inserts a TrFO Break Function, which then makes use of the stored RFCI values, in order to perform the required Iu UP protocol functions and interpret the payload. Further call scenarios for specific services that incur a TrFO break are described in chapter 6.

5.4.5 MGW Control Protocol Iu UP Package properties

The following is a summary of the Iu UP H.248 requirements; the procedures are valid for Iu UP in Support Mode:

Additional Package Properties:

Iu UP Termination Type: Values - Iu-RAN

- Iu-CN

Iu UP Initialisation Procedure: Values - Incoming

- Outgoing

Procedures:

Iu UP Initialisation procedure is always acknowledged between MGW peers. If a request for a Notification for the bearer establishment is requested then this shall not be sent until the acknowledgement for the Iu UP initialisation has also be returned.

The RFCI parameters are always stored against the MGW termination that received the Iu UP initialisation.

If a MGW has Iu UP termination property Initialisation Procedure = Incoming then it expects to received an Initialisation (either internally or externally).

If a MGW has Iu UP termination property Initialisation Procedure = Outgoing then it generates a network originated Initialisation PDU.

If a MGW has two terminations in the same context defined as supporting Iu UP package, then on receipt of an Iu Initialisation procedure from one side it shall forward the Iu UP initialisation procedure on to the peer MGW. This procedure shall be performed independently of the through-connection of the terminations in the context, but is dependent on the bearer connection from the other termination to its peer MGW being established.

If a MGW has one termination with Type = Iu-RAN and one with type Iu-CN in the same context then no forwarding of Iu UP initialisation out from the Iu-RAN termination shall be performed until an Iu UP initialisation has been received at the Iu-RAN side. If the RFCI values stored at the Iu-CN termination do not match the RFCI values stored at the Iu-RAN side then “RFCI Matching” may be performed to the Iu-RAN side – Iu UP initialisation is sent with the RFCI values from the Iu-CN side. No “RFCI Matching” is permitted at the Iu-CN side.

“RFCI Matching” may be delayed if terminations are not through-connected, triggered by connection modification otherwise it shall be performed immediately, this is implementation option

If “RFCI Matching” is not performed the MGW shall map the indexes for Iu frames from one side to the RFCI indexes from the other side.

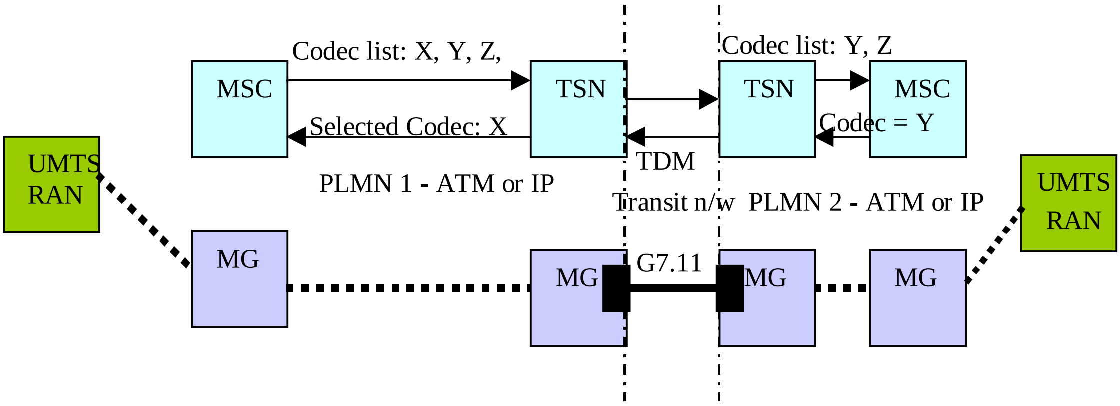
If a MGW has two Iu-RAN terminations connected to the same context then the “RFCI Matching” is performed to the termination latest defined.

If a MGW has two terminations with Iu UP package connected to the same context and both RFCI sets match then the MGW may switch into Iu UP transparent mode – no monitoring of the Iu frames is performed, provided that the terminations are through-connected.

If a H.248 procedure is received when a MGW is in transparent mode (but Iu UP is defined as support mode) that requires interpretation or interaction with the Iu UP then the MGW shall switch back to support mode, i.e. perform monitoring or termination of the Iu UP protocol.

5.5 TrFO/TFO Codec Negotiation Harmonisation

When OoBTC procedures are initiated to a node where compressed voice cannot be supported (either at the node or to the preceding node) then a transcoder is inserted. This can be due to the transport technology (e.g. TDM) or due to the



access technology (e.g. GSM). The OoBTC procedures can result in the following call scenarios:

Figure 5.5/1: Cascaded TrFO & Transcoding

In Figure 5.5/ 1 the OoBTC cannot proceed as the call crosses a transit network that does not support compressed voice. The same could occur if the transit network did not support out of band codec negotiation (Support in BICC is optional).

In Figure 5.5/2 the OoBTC procedures result in the call terminating to a GSM access. As the GSM radio access transcodes to default PCM codec, the OoBTC results in default PCM being the only codec that can be selected. The reply is passed back to the originating network, which then inserts a transcoder from default PCM to AMR for the UMTS radio access.

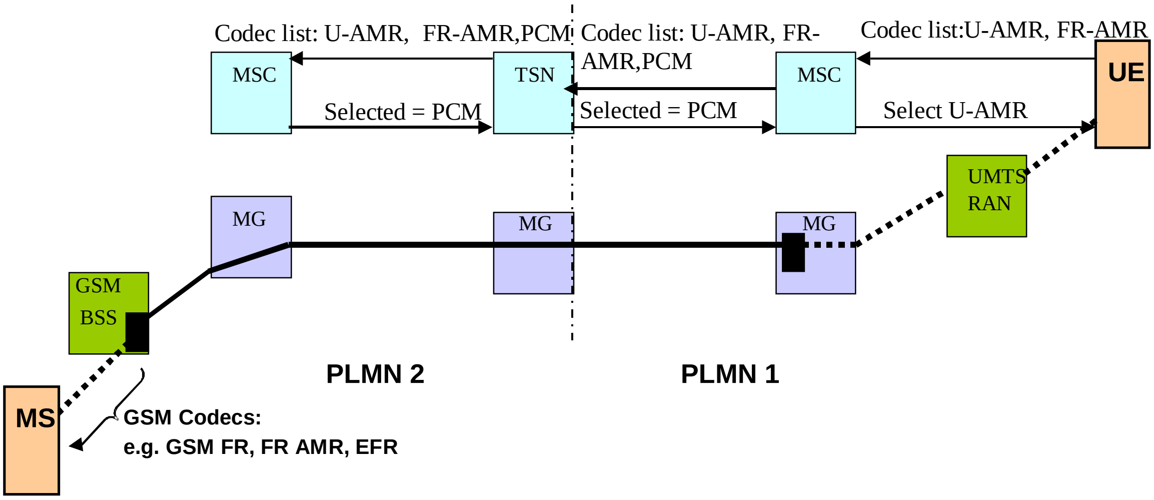


Figure 5.5/2: UMTS to GSM interworking

For TFO to establish between the transcoders in the above scenarios, each TRAU must send a codec list inband after the call has been established. If a common codec type is available (determined by pre-defined rules, described in TFO specification [10]) then the OoBTC procedures need to be informed so that a codec modification can be performed. This is shown in Figure 5.5/3. Note – a modification could also be required when a common codec type has been selected but the ACS is not common.

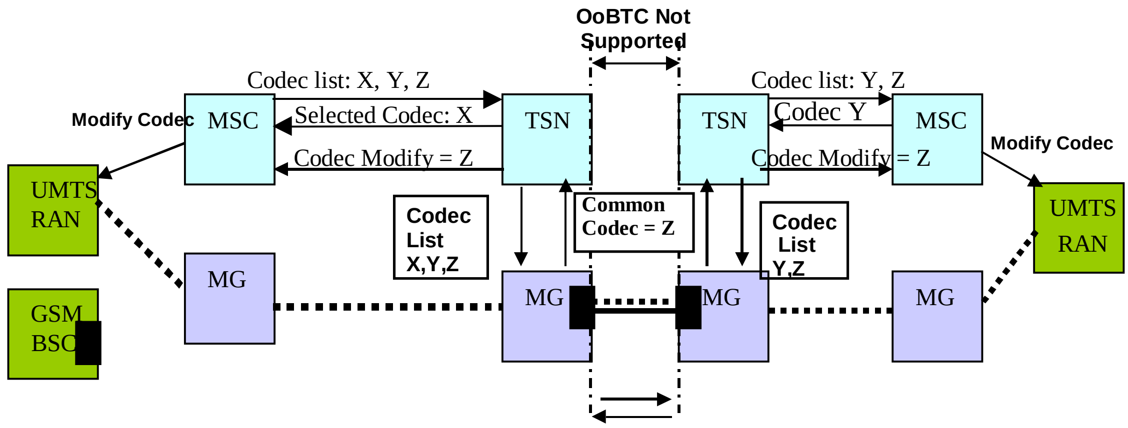


Figure 5.5/3: TFO support by OoBTC signalling

In H.248, the vertical MG control protocol, the coding types are specified by Media Stream Property, as defined by Annex C of H.248 specification. This stream property only allows one codec type to be specified. It does not permit a list of codecs to be sent to the MG. Further no property exists currently that defines the support of TFO. A TFO package for H.248 is defined to include the requirements for TFO.

The basic requirements are listed below:

i) Property for Codec List (same format as for [5])

ii) Event for common codec determined by TFO protocol

iii) Procedures to define TFO

5.6 CN Node handling of Codec Types & Codec Modes

The supported codec list received by the MSC in DTAP protocol [2] has no priority, whereas the list sent in the OoBTC procedures is sent with a level of preference. In order to support interworking with 2G systems it is recommended that MGWs support 2G EFR codecs and for GSM the FR AMR codec. In order to avoid modifications during handover between 2G and 3G systems the MSC nodes may give preference to a suitable 2G codec.

The originating CN node, while performing speech service negotiation with a terminating CN node, shall indicate the maximum number of modes that shall be selected during speech codec negotiation. This maximum number of supported modes may depend on optimisation strategies applied by the originating CN node.

The terminating CN node receiving this information compares the maximum number of modes received by the originating CN with its own one and shall decide on the minimum of both numbers to be applied as result of the negotiation.

The decision about the actual modes to be selected shall be left to the terminating CN node. In order to provide harmonisation of out of band codec negotiation (TrFO) and inband codec negotiation (TFO) the same codec selection mechanisms being defined for TFO shall be applied for TrFO. These rules shall be taken into account when forwarding a codec list from the originating node to proceeding node, both for TrFO and TFO.

When the MSC node requests a RAB assignment the Subflow Combinations provided shall either all be initialised by the RNC or all rejected with appropriate cause code.

The MSC shall always define “DTX” and “No Data” SDUs in addition to the negotiated speech modes. This is because for TrFO the RAB requested by one RNC must match that requested by the peer RNC – they are effectively the same RAB. If one MSC requires DTX support then the RAB requested by the far end MSC must also support DTX (even if it is not desired by that MSC). As no Out Of Band negotiation for DTX is supported nor DTX control to the UE, DTX shall be mandatory for TrFO connections.

5.7 Inband Rate Control

Inband rate control shall only allow the RNCs to set the maximum codec mode (maximum bitrate) from the set of codec modes that have been negotiated out of band. This procedure is called Maximum Rate Control. The final maximum mode selected results from a rate control request from one side and the maximum rate supported at the receiving side; the lower rate of these is selected. . This is known as Distributed Rate Decision. In TrFO maximum rate control shall be supported through the Iu UP protocol and through transit networks supporting compressed voice. The maximum rate control procedures are further defined within the Iu UP protocol [4].

When the MSC requests for a RAB to be assigned, it shall always define 1 speech mode SDU (lowest rate), DTX SDU and no data SDU as non-rate controllable. Other SDU formats for higher rates shall be defined as rate controllable.

At SRNS relocation the new RNC shall send a rate control frame at Relocation Detect indicating its current maximum rate, it will receive in the acknowledgement the current maximum rate from the far end. This procedure is called Immediate Rate Control. Again the distributed rate decision means both RNCs will operate within a common limit.

5.8 Modification Procedures

The OoBTC procedures shall support the following modification mechanisms:

i) modify Selected Codec (codec type or modes)

ii) modify Available Codec List (reduction of Available Codec)

The specific call flows when such procedures may be required are detailed in Chapter 6.

5.8.1 Modification of Selected Codec

In Figure 5.8.1/1 the basic codec modification procedure is shown. The principle is that the request for modification is made from one node through the network, each node with an MGW connection indicates to its MGW that a codec modification may occur, this also prepares the MGW for a bearer modification (based on the bearer requirements of the new codec). When the far end node is reached and the modification can be accepted, Modify Acknowledgement is returned and if the bearer must be increased then (as shown in the Figure 5.8.1/1, actions 10, 12, 14, 16) each MGW performs the required bearer modification. If bearer decrease is needed then no change to the bearer shall be made at this stage.

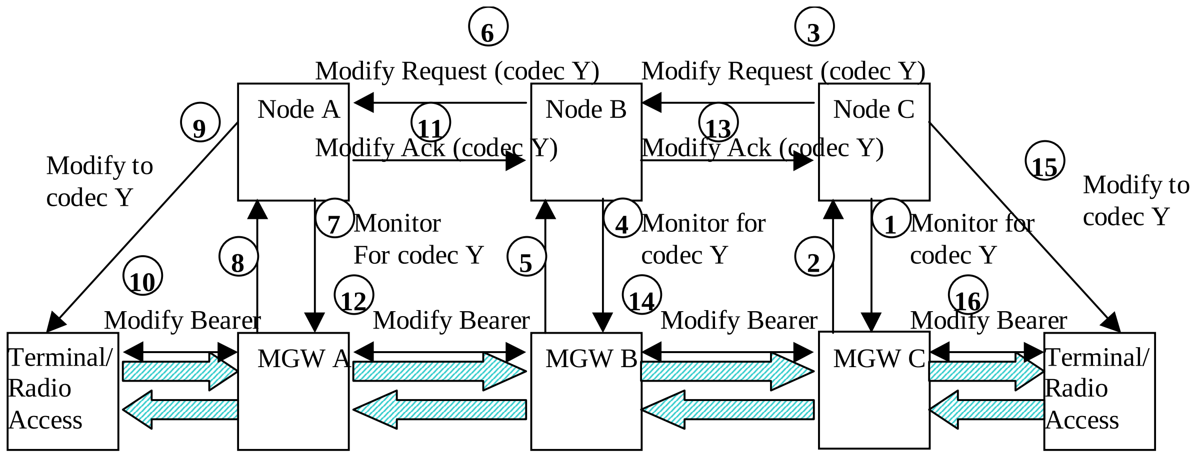


Figure 5.8.1/1: Codec Modification Control Procedures

When the Codec Modification initiating node receives the Modification Acknowledge, then it may order the change to the source codec. The MGWs are at this stage only monitoring for new codec type. As shown in Figure 5.8.1/2 the modification of the codec is performed as separate operation for Uplink and Downlink, this ensures that both the codec modification and bearer modification are synchronised.

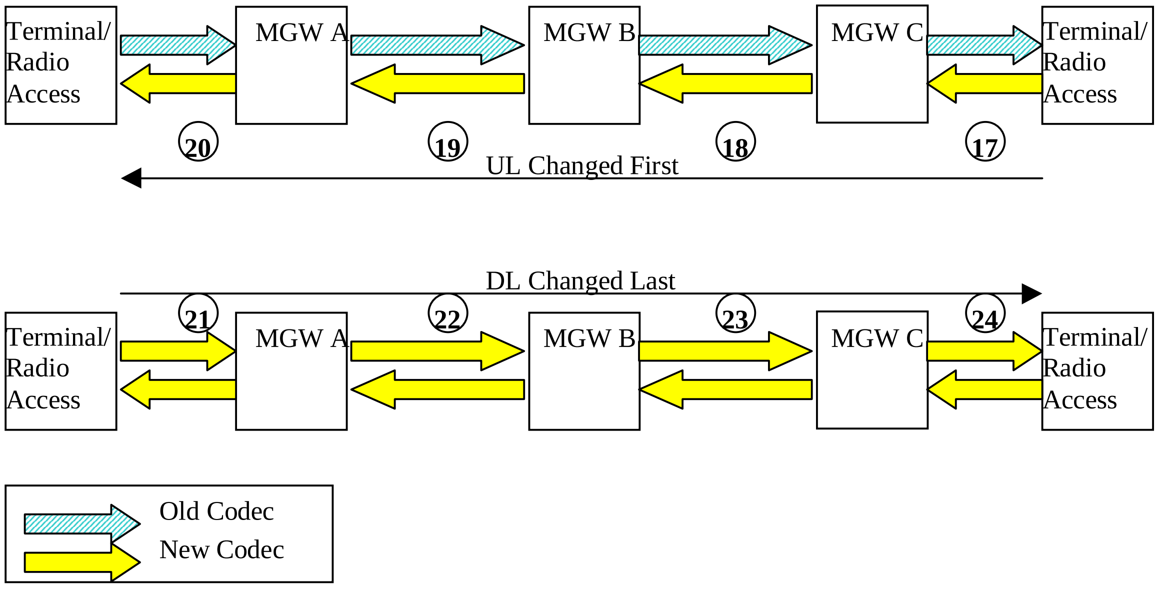


Figure 5.8.1/2: Codec Modification inband procedure

5.8.2 Modification of Available Codec List

Codec List modification may occur by “puncturing” of codec types or modes from the current Available Codec List. Note this shall not include puncturing of modes from the selected codec, as this would require Selected Codec modification. If a node performs a procedure (e.g. call forwarding) which results in a reduction to the list of Available Codecs then it shall send the new Available Codecs List to all preceding nodes indicating Codec List Modification.

5.9 DTMF Handling For TrFO Connections

DTMF from the UE is sent via DTAP procedures out-band. For a TrFO call the Originating MSC shall use an out-band DTMF procedure, all CN nodes shall support this procedure in their call control protocol. The out-band DTMF procedure shall also be used when TrFO is not achieved in order that TFO is possible. Insertion of DTMF in the PCM payload can result in the break of the TFO connection.

For terminating calls DTMF may need to be received by the core network (for voice-prompted services, voicemail control procedures etc). If the DTMF is received out-band then out-band procedures shall be maintained in core network.

If the DTMF is received for a TrFO call from an external network inband, in I.366.2 profile or RTP payload type, then the gateway MGW which interworks between Iu UP and the external framing protocol shall report the DTMF tones via H.248 procedures to its server. The server shall then use out-band procedures to pass the DTMF through the CN. See Figure 5.9/1.

The MGW may also optionally pass DTMF inband where such an option exists for the Nb interface, and is supported by the proceeding MGW.

Transcoding to default PCM to send DTMF tones shall be avoided for TrFO connections.

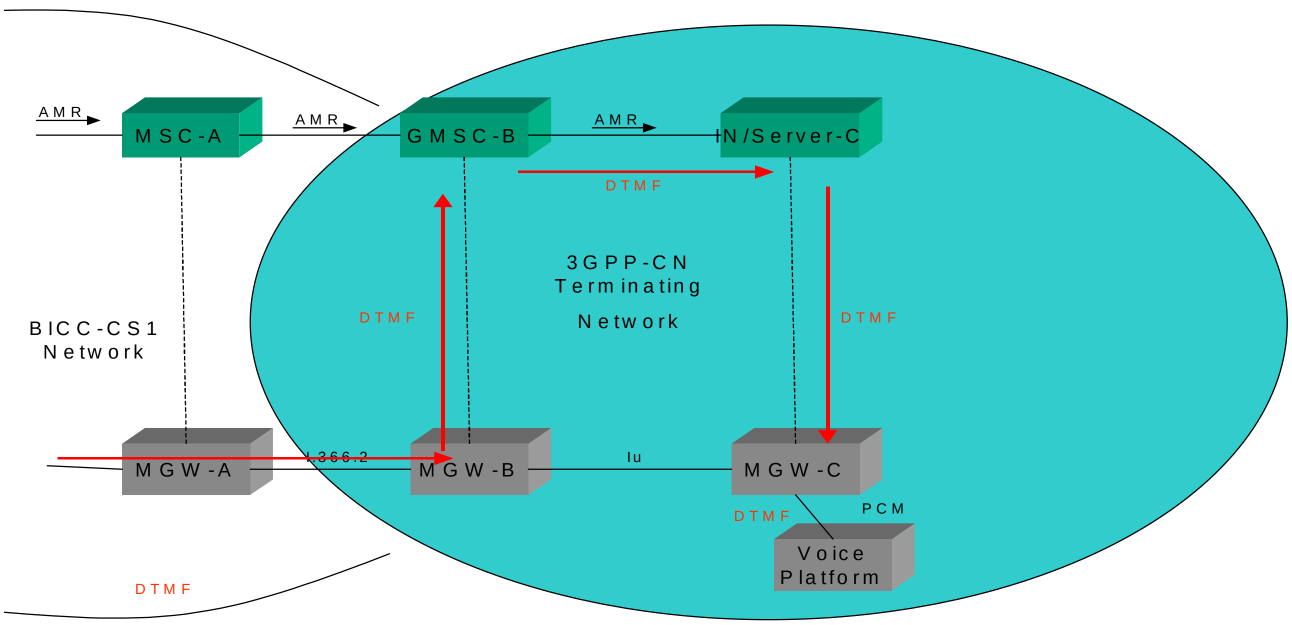


Figure 5.9/1:DTMF received inband from external network

6 Detailed Call Procedures

6.1 Mobile to Mobile TrFO Call Establishment

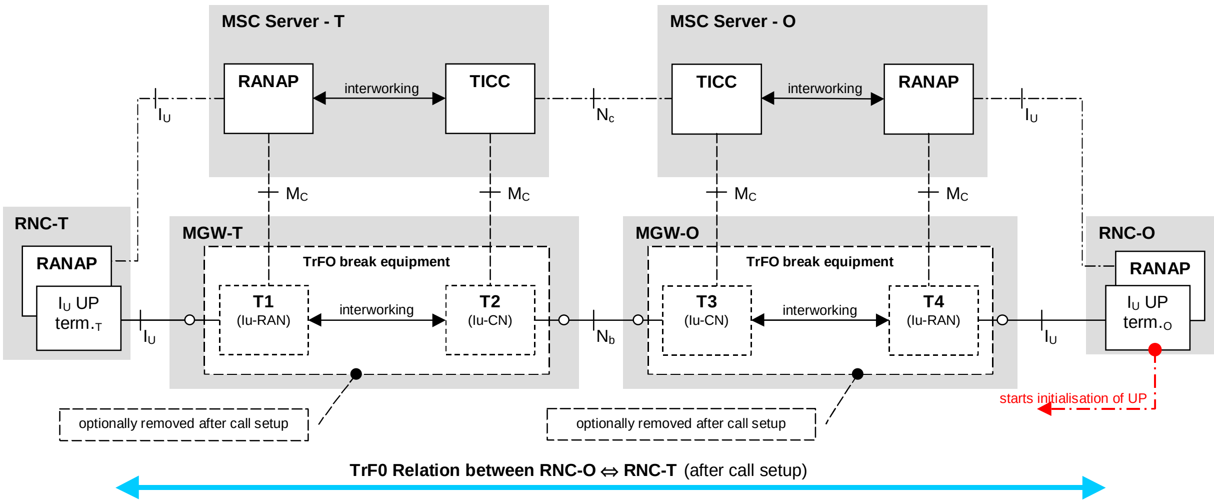


Figure 6.1/1: Configuration during Call Setup of a Mobile to Mobile Call.

Following network and protocol entities are involved in the scenario, outlined in Figure 6.1/1:

RNC-T, **RNC-O**: terminating/originating RNCs

MSC Server-T, MSC Server-O: MSC Servers, performing service, i.e. codec negotiation

MGW-T, **MGW-O:** terminating/originating MGWs with the optional capability to insert/remove so called

TrFO break equipment: (**TBE**s), i.e. contexts containing an UTRAN- and a CN side I_U UP termination (**T1 – T4**), inter-working in a distinct manner on control level. [Note: *context* is meant to be the H.248 specific throughout the document]. It is aimed to design protocols for TrFO in a way, that these pieces of HW can be removed after call setup phase to allow to revert to “simple” AAL2 switching in case of ATM transport.

I_U UP term.$_T$, I_U UP term.$_O$: Terminating- and originating-side TrFO peers (I_U UP terminations in RNC’s in Figure 6.1/1)

RANAP, **TICC**:C-plane protocol incarnations, responsible for codec negotiation, controlling the respective interfaces (I_U, N_C), creating, modifying, removing etc. terminations and contexts.

The final configuration is (at least logically) an end to end TrFO relation between RNC-T and RNC-O with the option to remove the TBEs from the user data path, i.e. to revert to pure AAL2 switching in case of ATM Transport.

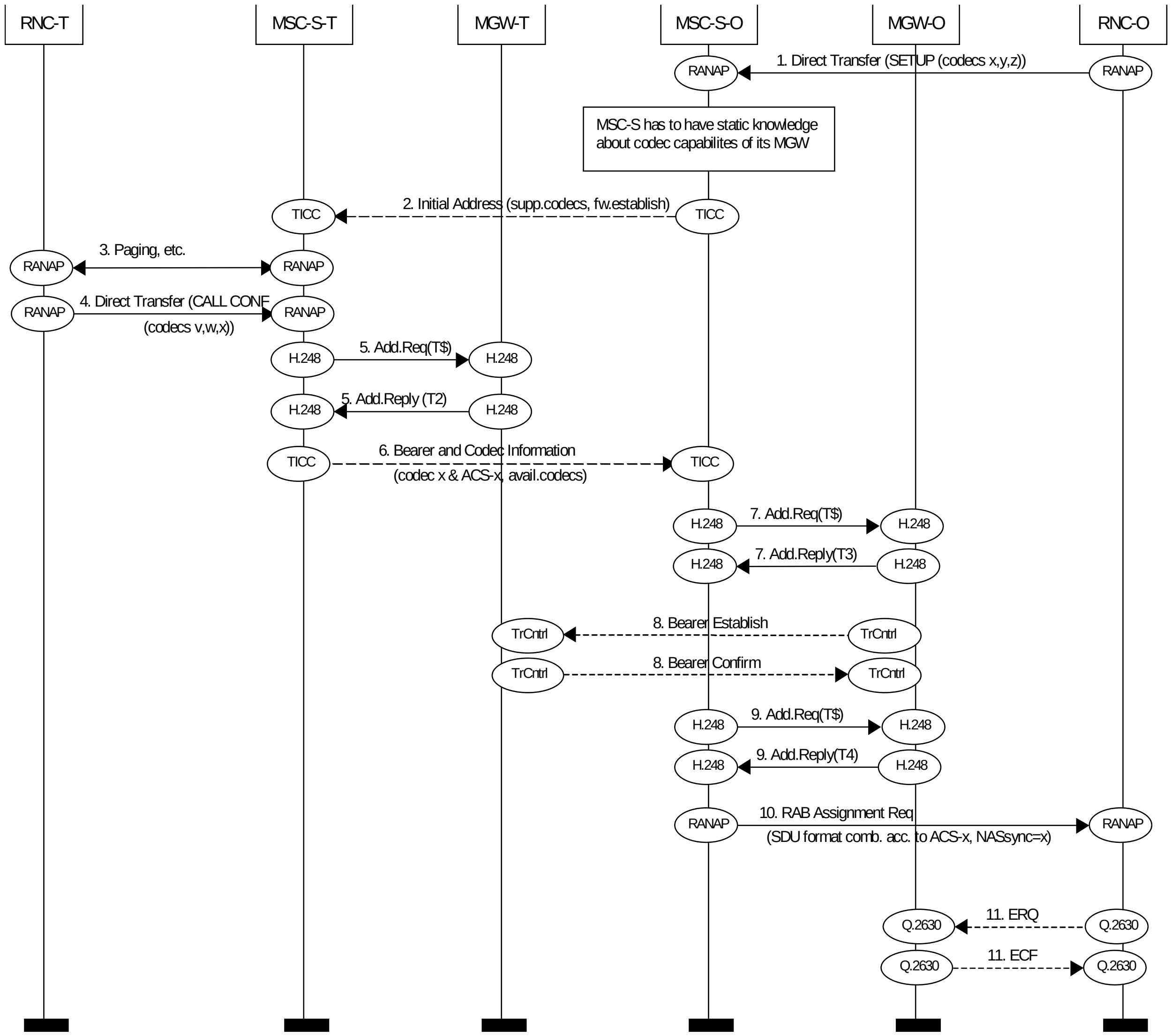


Figure 6.1/2: Call Setup. Mobile to Mobile Call. Message Flow part 1.

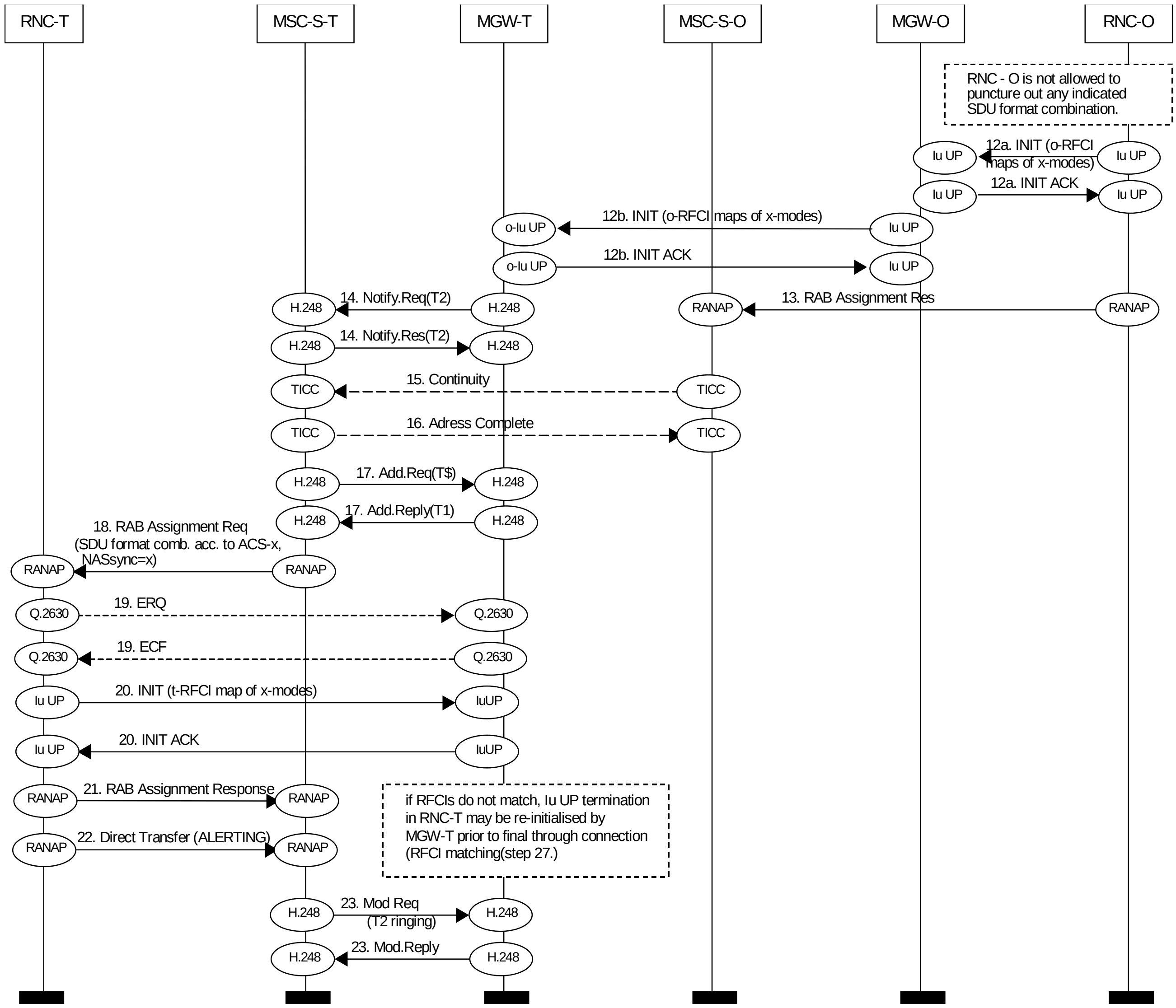


Figure 6.1/3: Call Setup. Mobile to Mobile Call. Message Flow part 2.

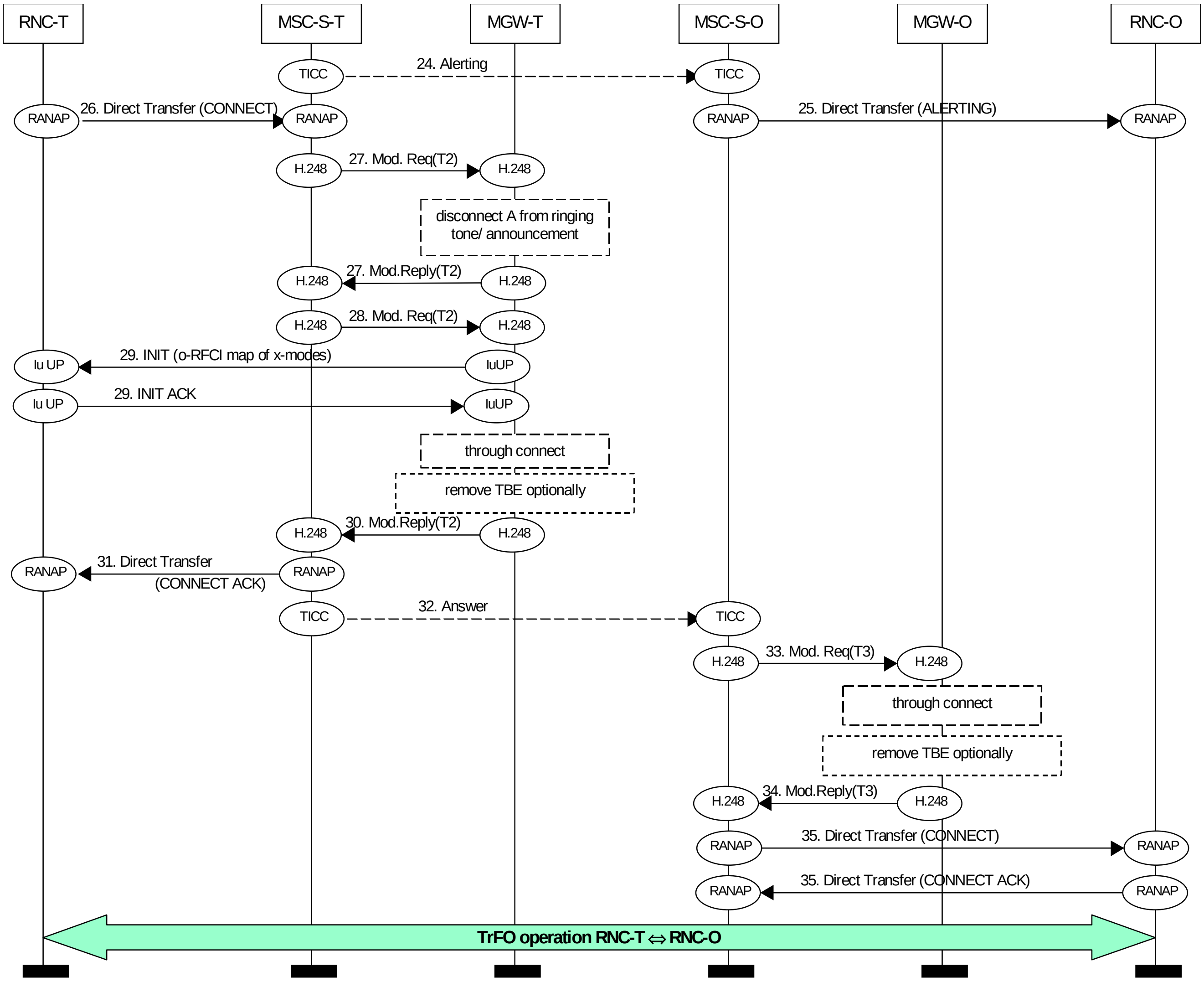


Figure 6.1/4: Call Setup. Mobile to Mobile Call. Message Flow part 3.

Codec negotiation

Step 1. to 6. gives the codec negotiation phase. The mobiles inform the network about their capabilities (1. and 4.). Afterwards the MSC-Server performs codec negotiation according to chapter 5.6.

Network side bearer establishment

MSC-T/MSC-O shall request seizure of network side bearer terminations with IuUP properties (see steps 5. and 7.). Intermediate CN nodes that may perform certain service interactions (e.g. IN nodes) have to seize terminations with IuUP properties as well.

RAB Assignment

RAN side terminations with IuUP property have to be seized (9. and 17.) before sending RAB Assignment (steps 10. and 18.), that contains RAB parameters according to the selected codec and the negotiated ACS. In addition, the respective NAS synchronisation indicator shall be included.

6.2 SRNS Relocation during TrFO

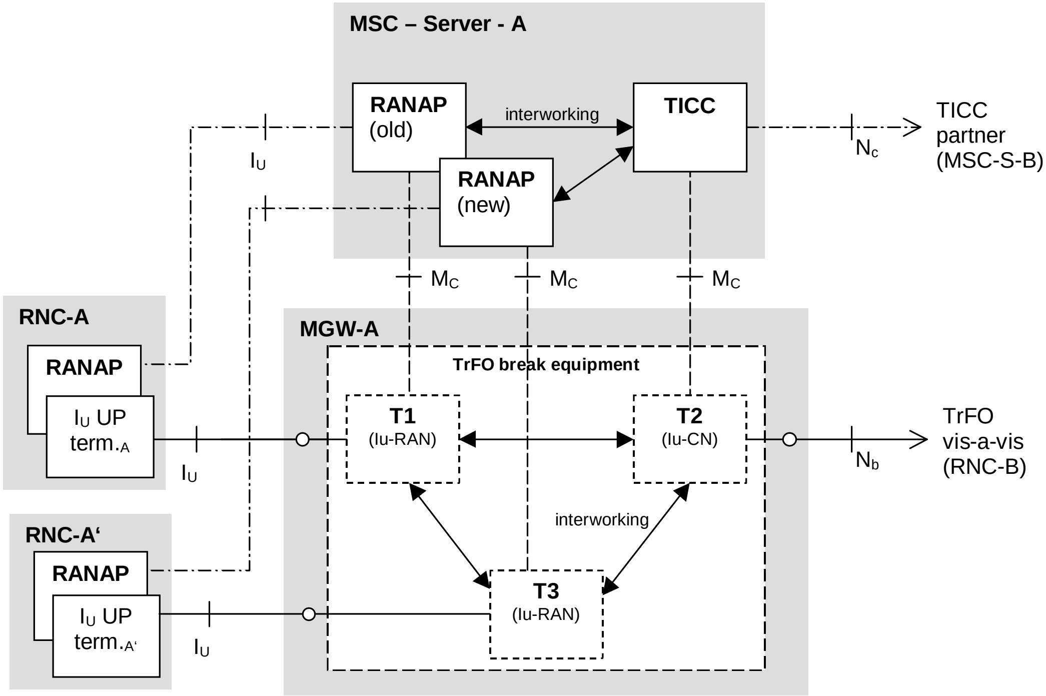


Figure 6.2/1: Configuration during SRNS Relocation

Figure 6.1/1 shows the configuration during relocation. After setting up the new I_U interface (towards RNC-A') until releasing the old one, the original TrFO relation (A⇔B) and the target TrFO relation (A'⇔B) exist in parallel. Within the respective context (TBE) interworking between T1, T2 and T3 is necessary:

T3 will perform initialisation towards RNC-A'.

T2 will hide initialisation performed on $I_{U,A'}$ from RNC-B.

If the option to remove the TBE was applied after call setup, the whole context (TBE) needs to be inserted prior to performing SRNS Relocation. Initialisation data need to be available within MGW-A. After Relocation, the context (TBE) may be removed again, i.e. the MGW-A again acts as a pure AAL2 switch.

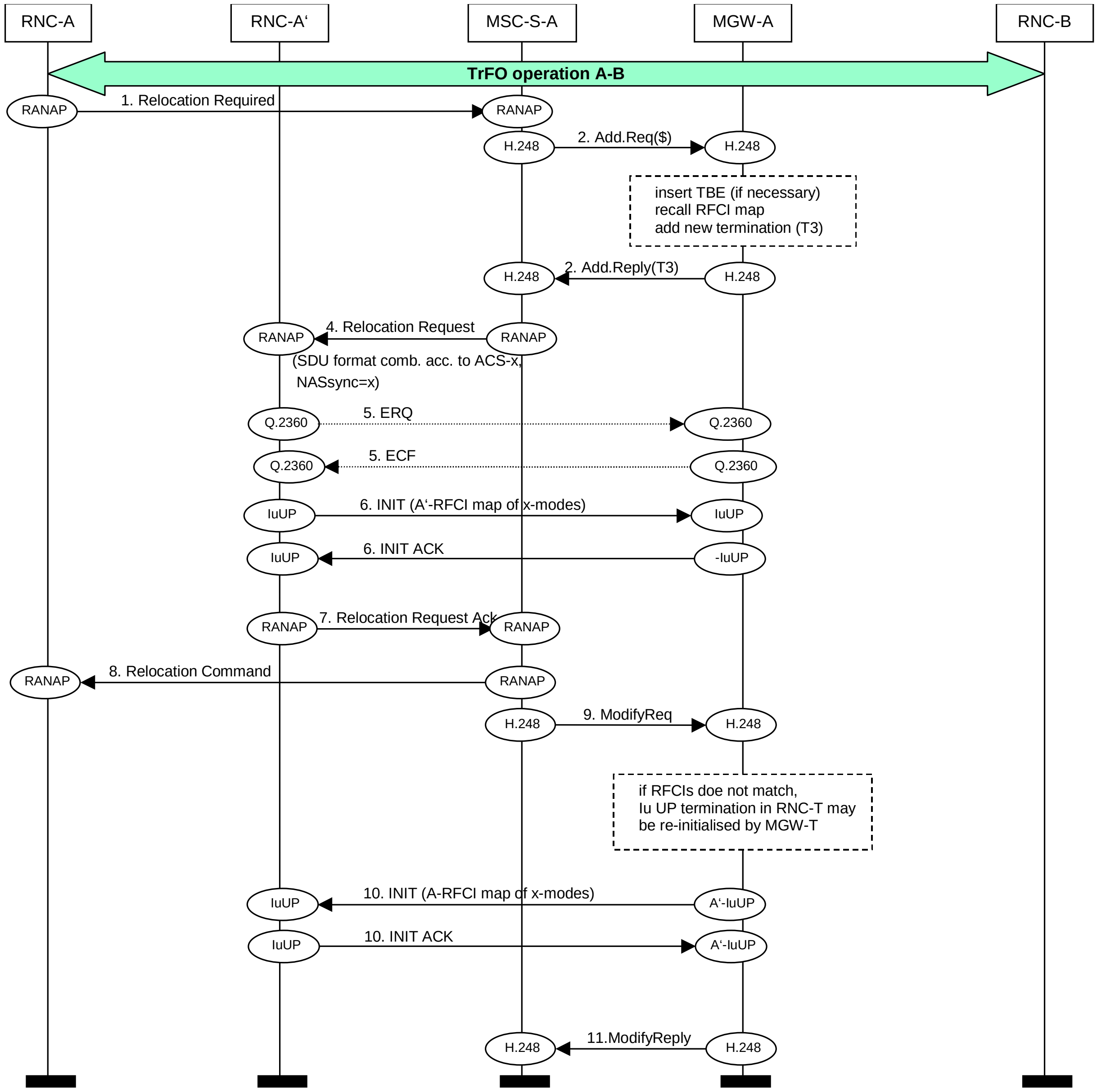


Figure 6.2/2:SRNS Relocation and TrFO. Flow chart part 1.

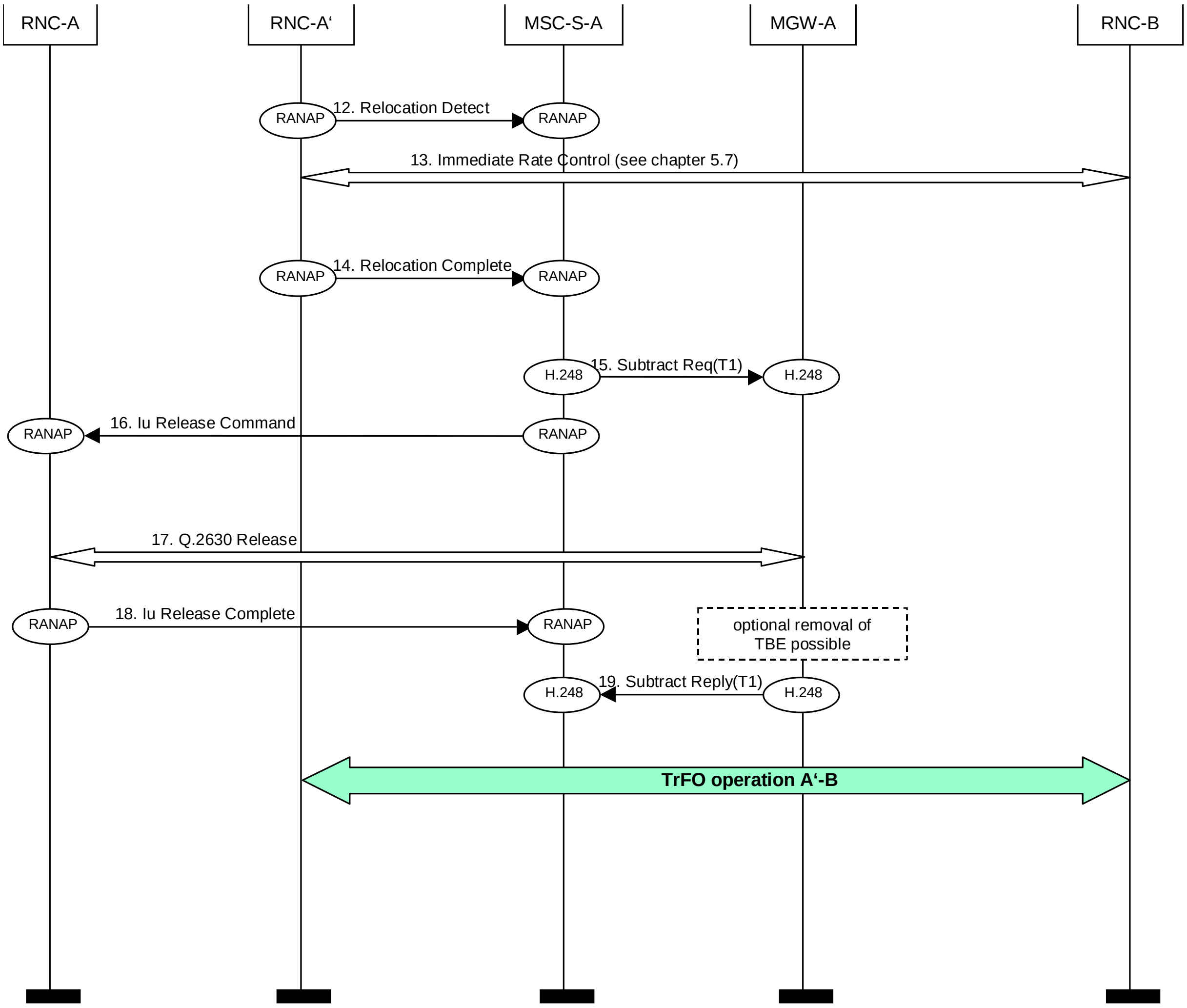


Figure 6.2/3 SRNS Relocation and TrFO. Flow chart part 2.

RAB Assignment on the new Iu leg:

A RAN side terminations with IuUP property (T3) has to be added to the already seized call context (step 2.) before sending Relocation Request (4.), that contains all the RAB parameters already applied on the Iu leg towards RNC-A.

UP initialisation

RNC-A' shall accept the requested set of codec modes and is not allowed to puncture out any negotiated mode. The INIT frames shall be according to the RAB parameters received.

At reception of an INIT frame from the new RNC, the termination at MGW-A shall not perform forwarding of the IuUP initialisation. The MGW shall check whether the received RFCI allocations match the stored RFCI allocation. If it does not match, it may re-initialise the IuUP towards RNC-A' at this point in time.

Removal of TrFO Break Equipment (TBE)

If the MGW supports the removal of TBEs, it shall insert the TBE before seizing the additional termination. It may again remove the TBE after performing RFCI matching and through-connection of the new termination and the termination to the far end party.

6.3 IN and Call Forward SS

In some cases, IN services (e.g. voice prompting) are triggered at CC-IN nodes that require the establishment of an UP bearer for tones or announcements to be sent to the calling party. In order to establish this bearer, it is necessary that the CC-IN node temporarily selects one codec from the codec list sent from the initiating node, and informs the initiating node about the selected codec. Afterwards, the call may continue its establishment to the another node, which may not support the selected codec but requests that another one in the list be selected instead.

A similar situation arises with the CFNRy supplementary service. A UP connection needs to be established between the originating and "provisional" terminating CC nodes to enable ringing tones to be sent to the calling party. The type of codec must be agreed prior to the establishment of the bearer connection. Afterwards, the call is redirected to another node that may not support the selected code but requests the selection of another one.

6.3.1 TrFO interworking with SS (VMSC = service interworking node)

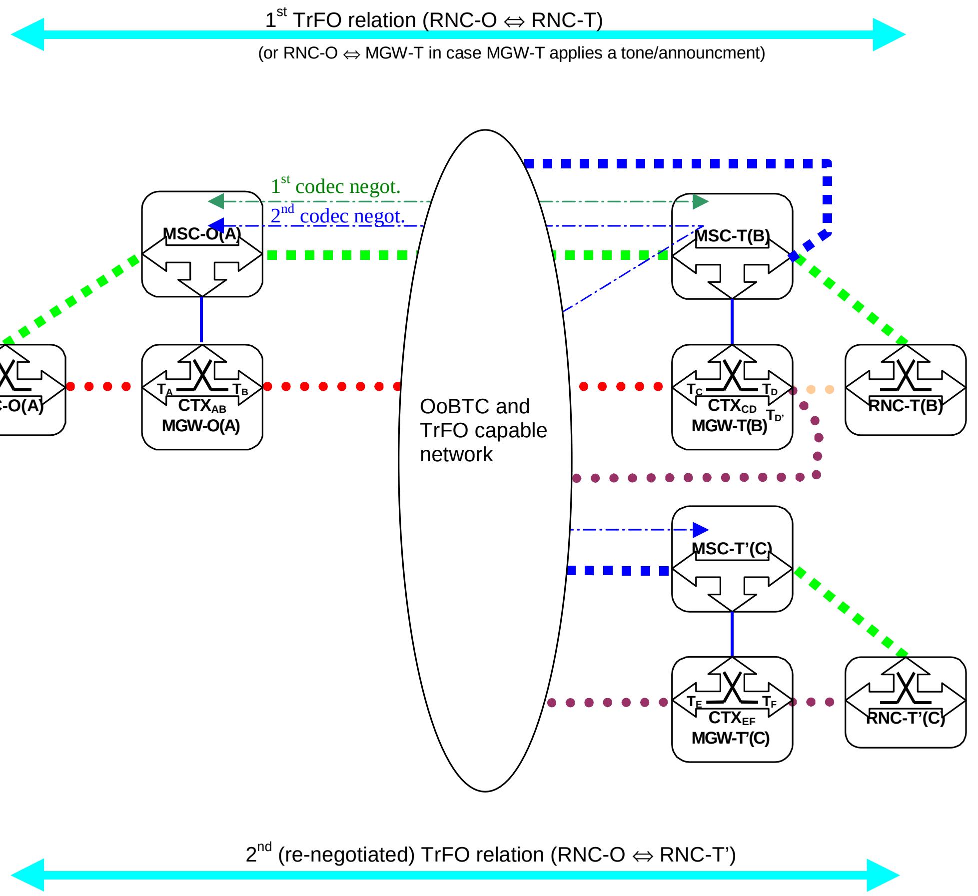


Figure 6.3.1/1. Codec Modification in case of SS interworking

In case of supplementary service interworking, it may become necessary to apply codec modification out of band. Figure 6.3.1/1 shows the network model, that may apply for a certain set of SS's (call deflection (CD), call forwarding on no reply (CFNRy), CF on user determined busy (CFUB), etc.). Common to these scenarios is:

- the service interworking is controlled by the VMSC (this is common to all SSs).
- MSC-T extends the call towards MSC-T' according to the forwarded-/deflected-to-number.

An intermediate TrFO relation will in general already exist between two RNC's (RNC-O and RNC-T in figure 6.3.1/1) before the call is diverted to another node, as the ringing tone was applied in backward direction.

In order to perform codec negotiation with the third node (MSC-T') as well it is necessary to forward the supported codec list from MSC-O. MSC-T' signals back the codec it selected and the available codec list. MSC-O is able to detect whether UP re-initialisation and bearer modification is required. Further call handling follows the mobile to mobile call establishment (see section 6.1).

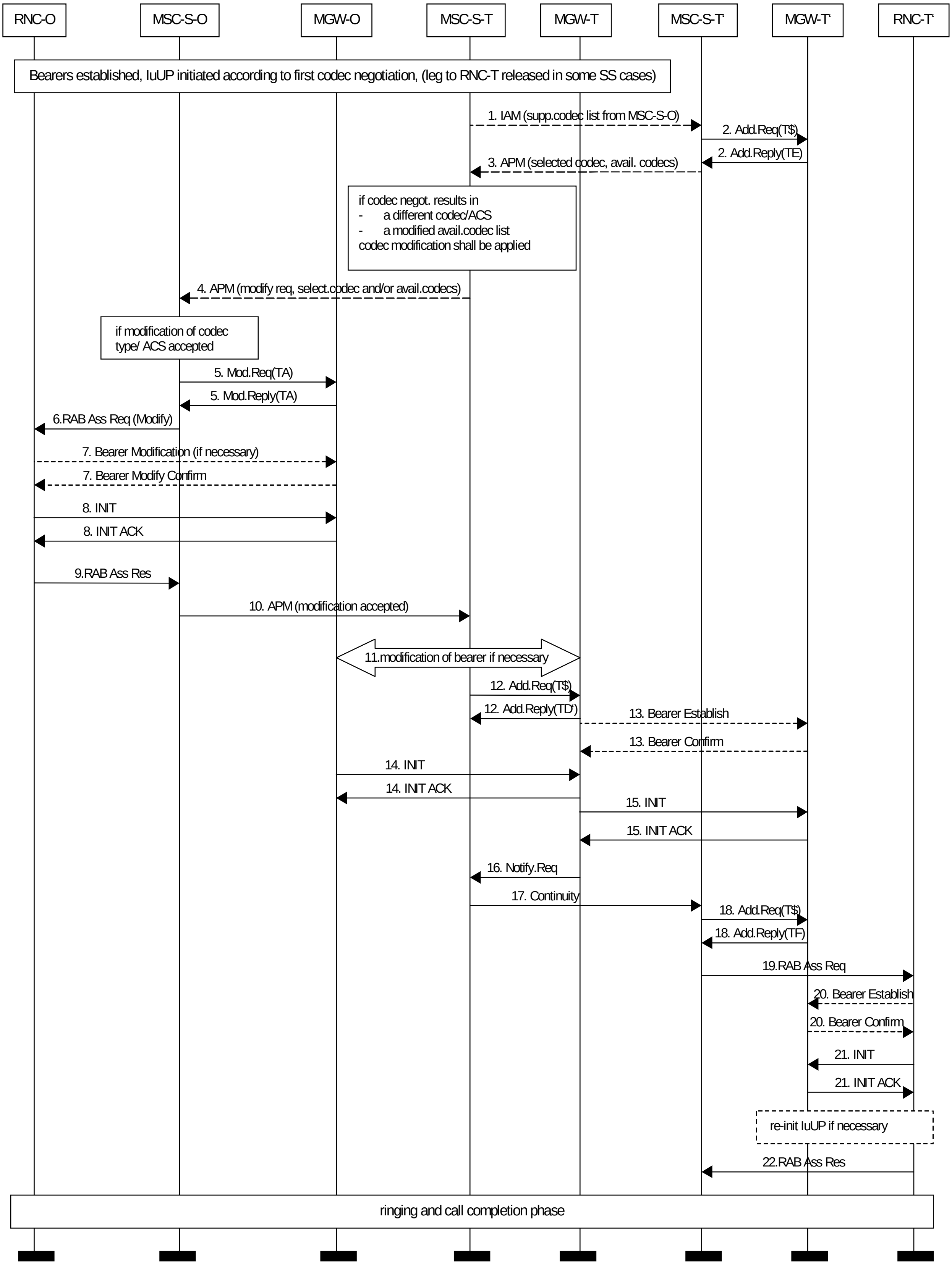


Figure 6.3.1/2: Codec Modification for SS-interworking & UP re-initialisation.

6.3.2 IN interworking (VMSC ≠ service interworking node)

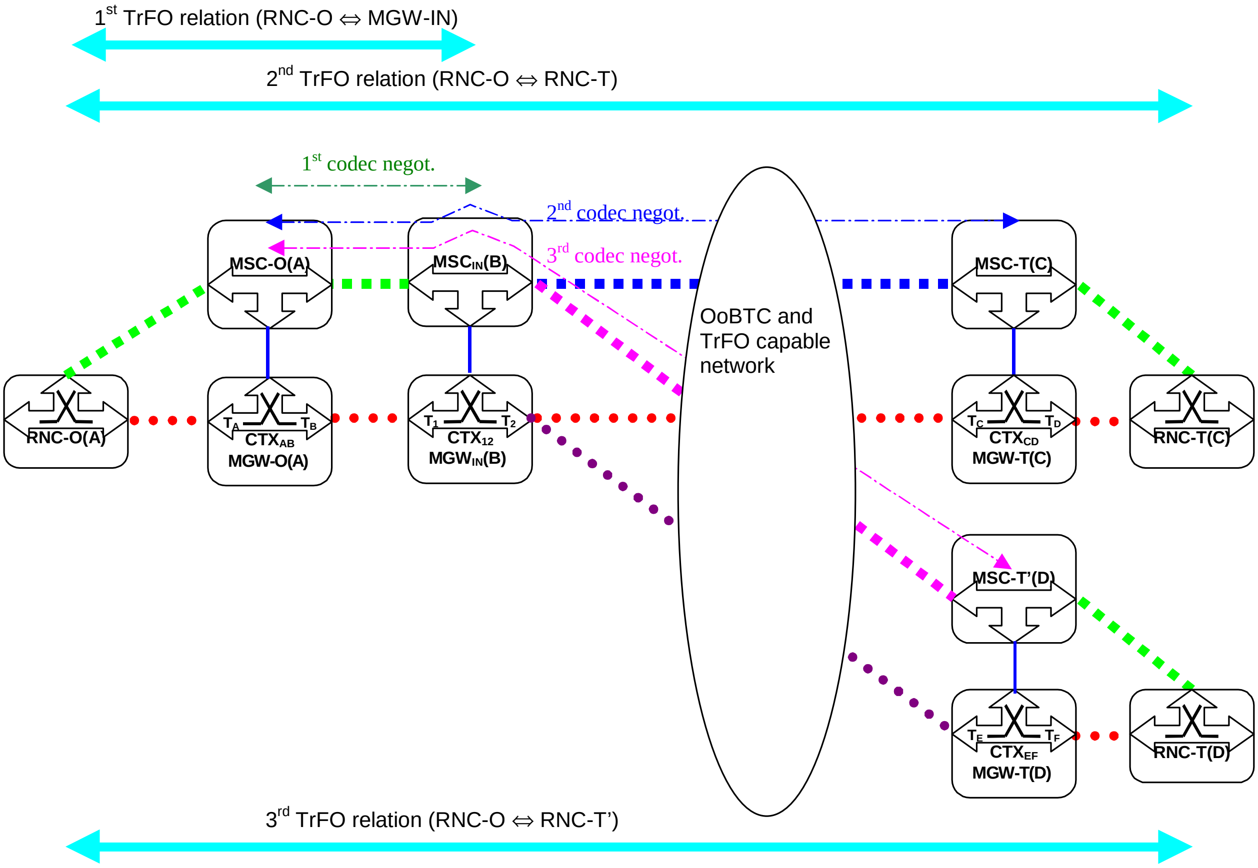


Figure 6.3.2/1. Codec Modification in case of IN interworking

Common to IN interworking scenarios is that service interworking is controlled by an IN service node that is generally not the VMSC.

IN interworking (i.e. in case of a separate IN service node, this is often a Gateway-MSC) may interrupt call establishment and apply an intermediate announcement back to the originating side. This means, that codec negotiation was in fact performed between the IN service node and the MSC-O.

When performing further call establishment, it is necessary to proceed with codec negotiation towards MSC-T. The codec negotiation process shall consider the capabilities of MGW-IN.

IN services, similar to call forwarding SS, are possible. The fact that this service interworking is controlled by an IN service node, may cause, that the leg towards MSC-T has to be released and a new leg towards MSC-T' will be established. Codec negotiation is again necessary from MSC-IN on.

The sequence chart given in figure 6.3.1/2 applies in principle for the 1st and the 2nd negotiation scenarios with following modifications:

- as MSC-IN may be involved in subsequent service interworking again, the capabilities of MGW-IN shall be taken into account during codec negotiation with MSC-T or MSC-T'. This means, that the codec list forwarded to the succeeding nodes is in fact the available codec list of the 1st negotiation.
- For the 3rd negotiation scenario, the leg between MSC_{IN}(B) and MSC-T (C) has to be released and a new leg toward MSC-T'(D) has to be setup.

6.4 Information flow for interaction with Multiparty SS

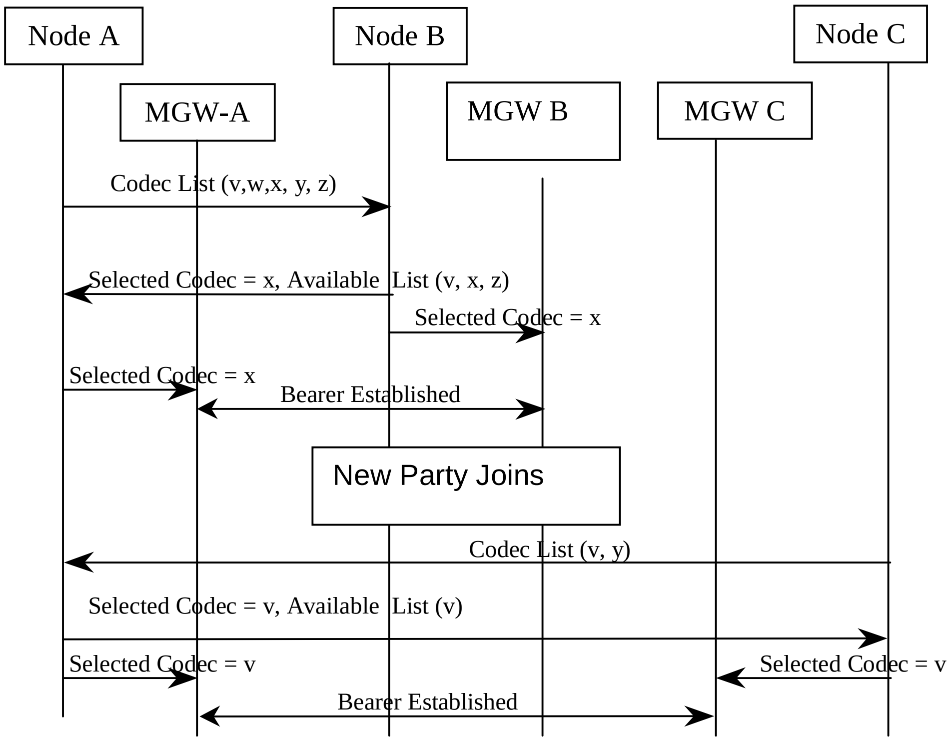


Figure 6.4/1: Multi-party Call

The operation of the MGW for conference calls is implementation dependent. The sequence in Figure 6.4/1 shows three connections to the MGW, where two were configured TrFO and have matching codecs but the third connection could not be made with the same codec type.

The Iu UP connections for each multi-party call leg shall be terminated in the MGW where the multi-party call is controlled. The MGW shall control each connection independently during the multi-party call.

When the multi-party call is released, if two parties remain in the connection it shall be possible to either revert directly to a TrFO connection if both codecs match or OoBTC procedures could be performed to modify one or both of the codec types to achieve a TrFO connection. However, if the Server does not perform this then the MGW shall continue to resolve the difference in codecs by internal transcoding procedures.

6.5 Information flow for handover from UMTS to GSM after TrFO establishment

Inter-system handover procedures are described at call control level in [11] and details for bearer independent architecture is described in [8]. For TrFO connected UMTS call, when a handover occurs to GSM radio access, by definition the A-interface to the BSC shall be default PCM. Prior to receipt of Handover Detect the Anchor MGW has one leg (A-interface) stream mode as default PCM and two terminations with compressed voice codec properties. It is recommended that after the Handover Complete procedure, the network property is maintained as compressed. Thus the Anchor MGW inserts a “TFO Partner” transcoder. Thus no modification to the compressed bearer to 64k PCM is required. TFO procedures may then ensure that speech quality is maintained by avoiding transcoding.

In the Intra-MSC case shown in Figure 6.5/1 the MSC controlling the handover has both codec lists for each radio access. The codec negotiation for the UMTS call was performed end to end with UMTS list. If this negotiation resulted in a codec being selected that is also included in the GSM list then at handover the MSC shall indicate this codec as the current speech version to the BSC and TFO can be achieved. If the selected codec is not supported for the GSM radio access but the GSM list contains a codec that is also in the Available Codecs list then the MSC has the option to perform codec modification to ensure TFO can be achieved. The MSC may also perform codec list modification by sending forward the GSM list to update nodes in the network of the change to the Available Codecs List.

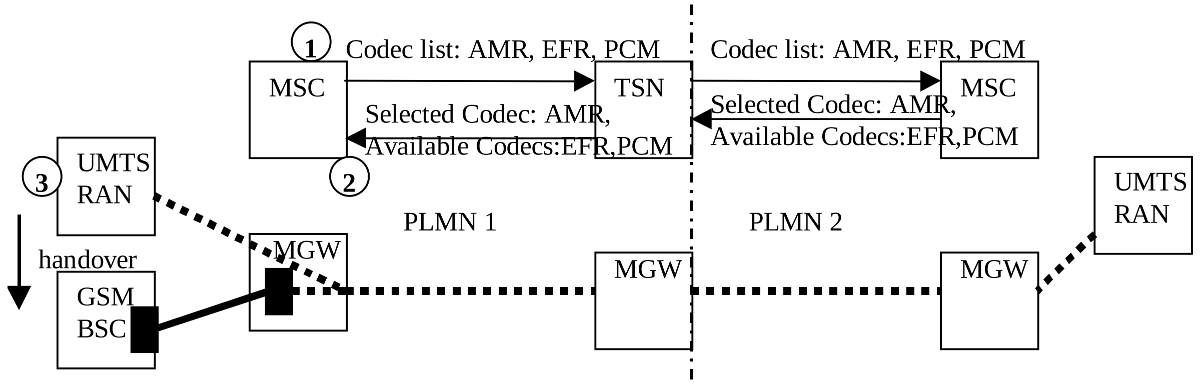


Figure 6.5/1: UMTS to GSM Inter-System Handover

If the Inter-system handover is an inter-MSC handover then the Anchor MSC sends the current speech version and the supported speech versions in the Prepare Handover Request message to the MSC-B. If the current speech version (codec selected for UMTS call) is not included in the GSM list then the MSC-A shall indicate a preferred codec in the current speech version parameter. The speech version for the GSM access that is finally selected by the MSC-B's BSS, is returned to MSC-A in the Prepare Handover Response message. The MSC-A can then decide if codec modification or codec re-negotiation shall be performed as described for the intra-MSC case. The MSC-B shall always assume default PCM across the E-interface, as there is no possibility to perform codec negotiation prior to performing the handover. The connections are shown in Figure 6.5/2.

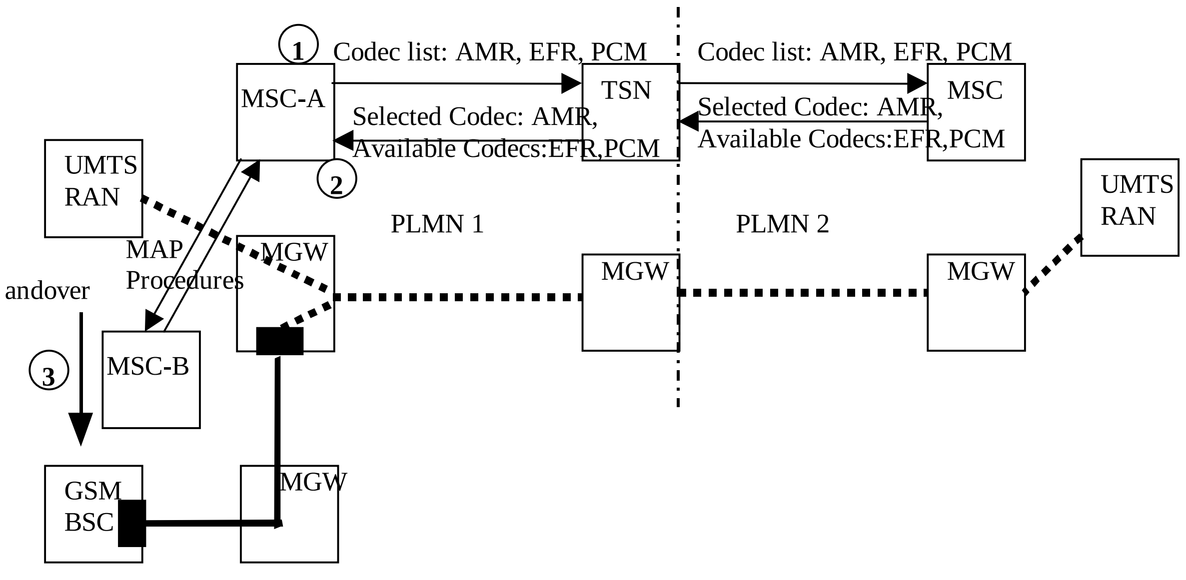


Figure 6.5/2: Inter-MSC, UMTS to GSM handover

6.6 Call Hold/Call Wait

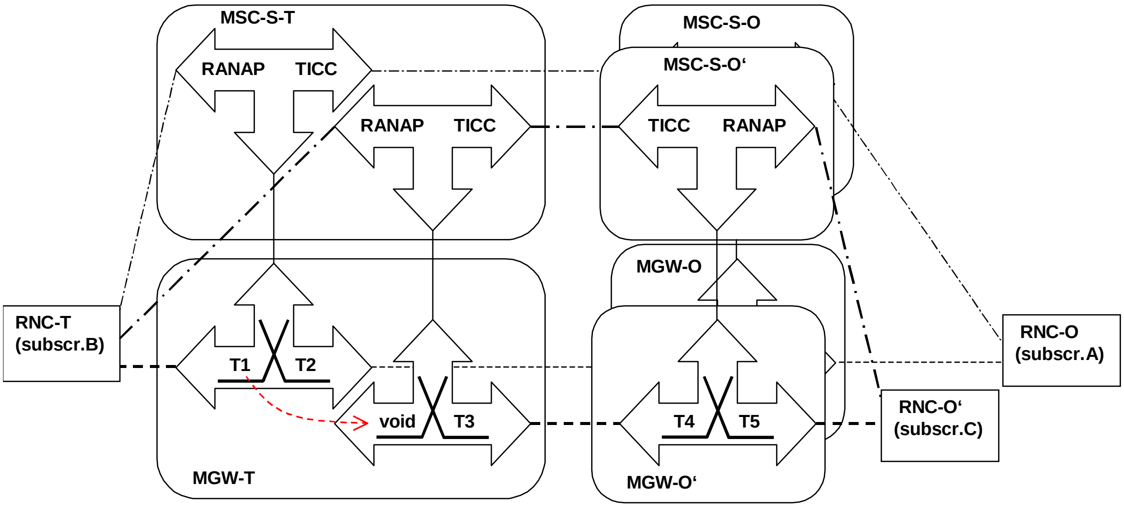


Figure 6.6/1: Configuration during Call Hold / Call Wait scenario

This scenario assumes subscriber C (served by RNC-O’) calls subscriber B (served by RNC-T), currently in communication with subscriber A. Subscriber C receives a tone/announcement, applied by terminating side. Then subscriber B puts subscriber A on Hold and A receives an announcement (applied again by terminating side.)

MGW-O has to establish an originating side call context (T4, T5), MGW-T the respective terminating one (T3 only, T1 from subscriber will be moved to it during the scenario), the B party context has to be inserted into path again (if TBE was removed).

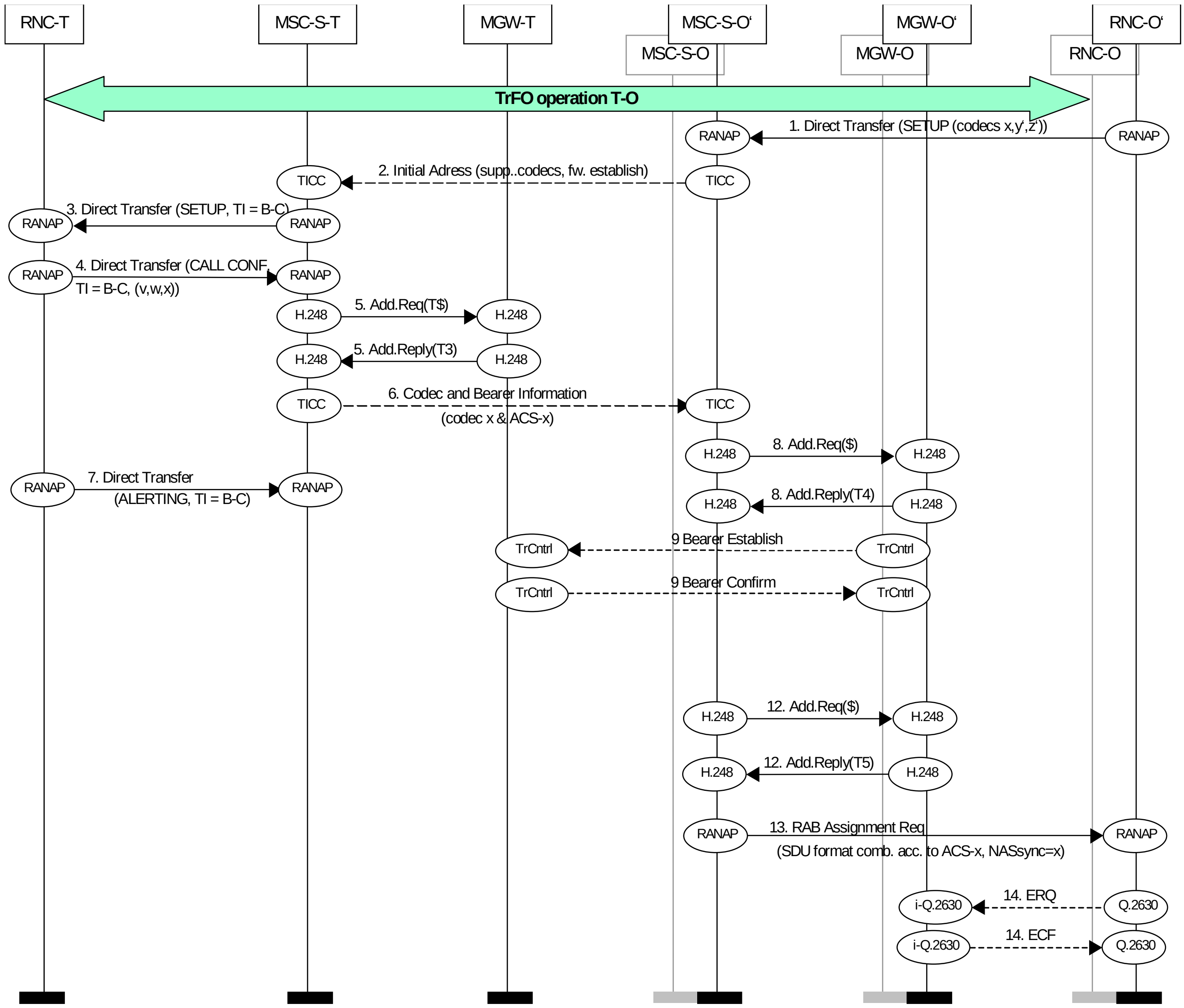


Figure 6.6/2: Call Hold/Call Wait and TrFO. Message flow part 1.

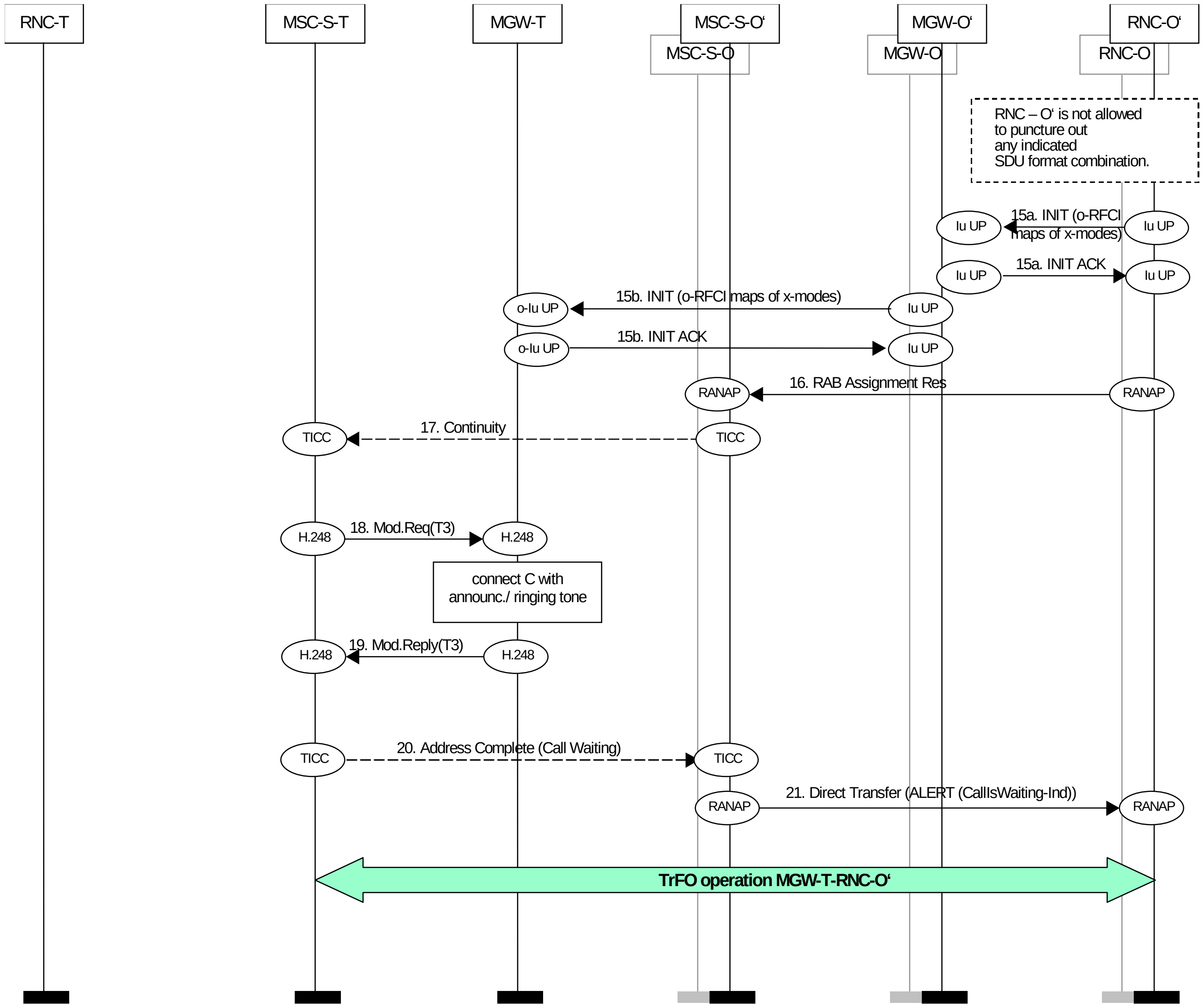


Figure 6.6/3: Call Hold/Call Wait and TrFO. Message flow part 2.

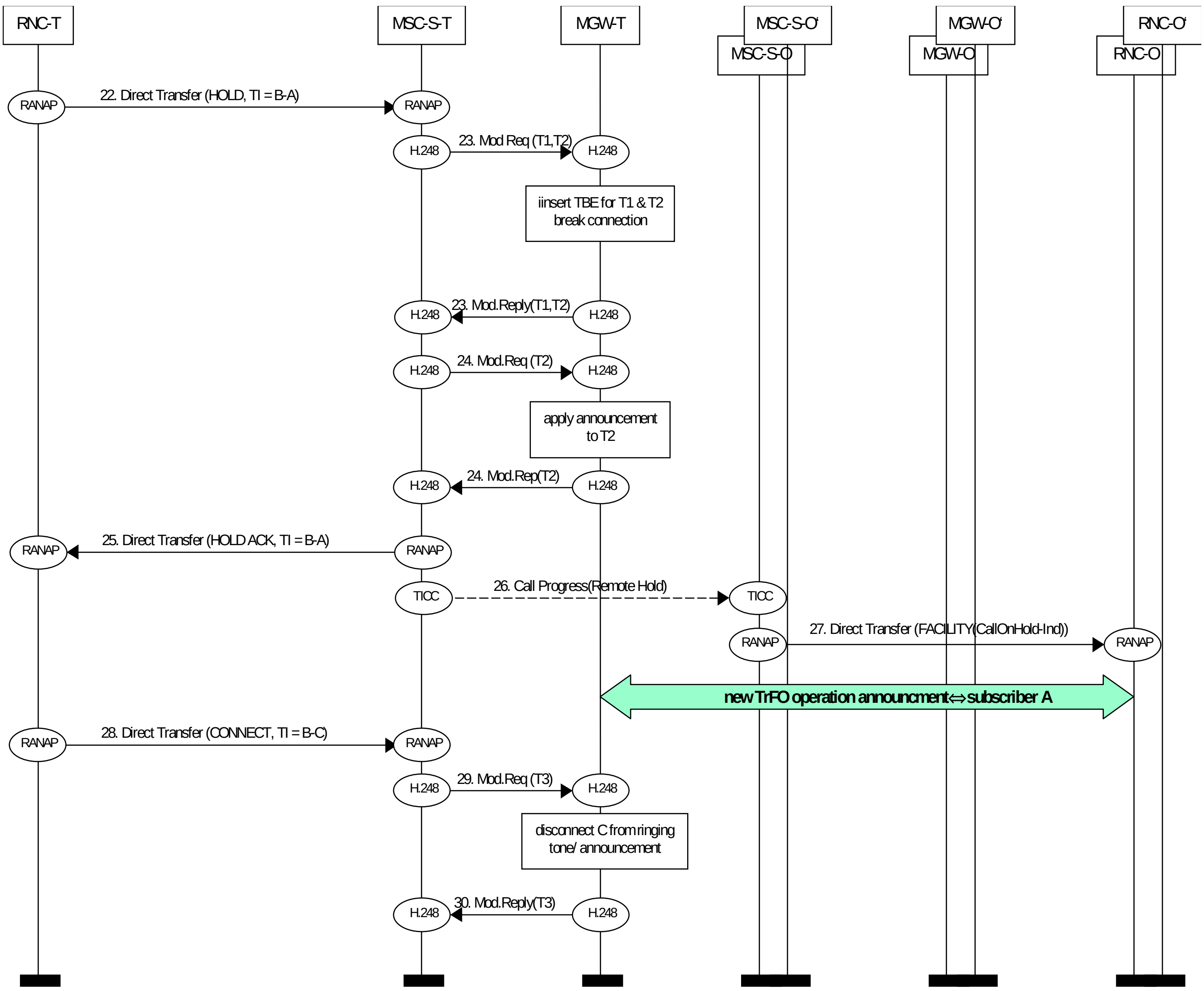


Figure 6.6/4: Call Hold/Call Wait and TrFO. Message flow part 3.

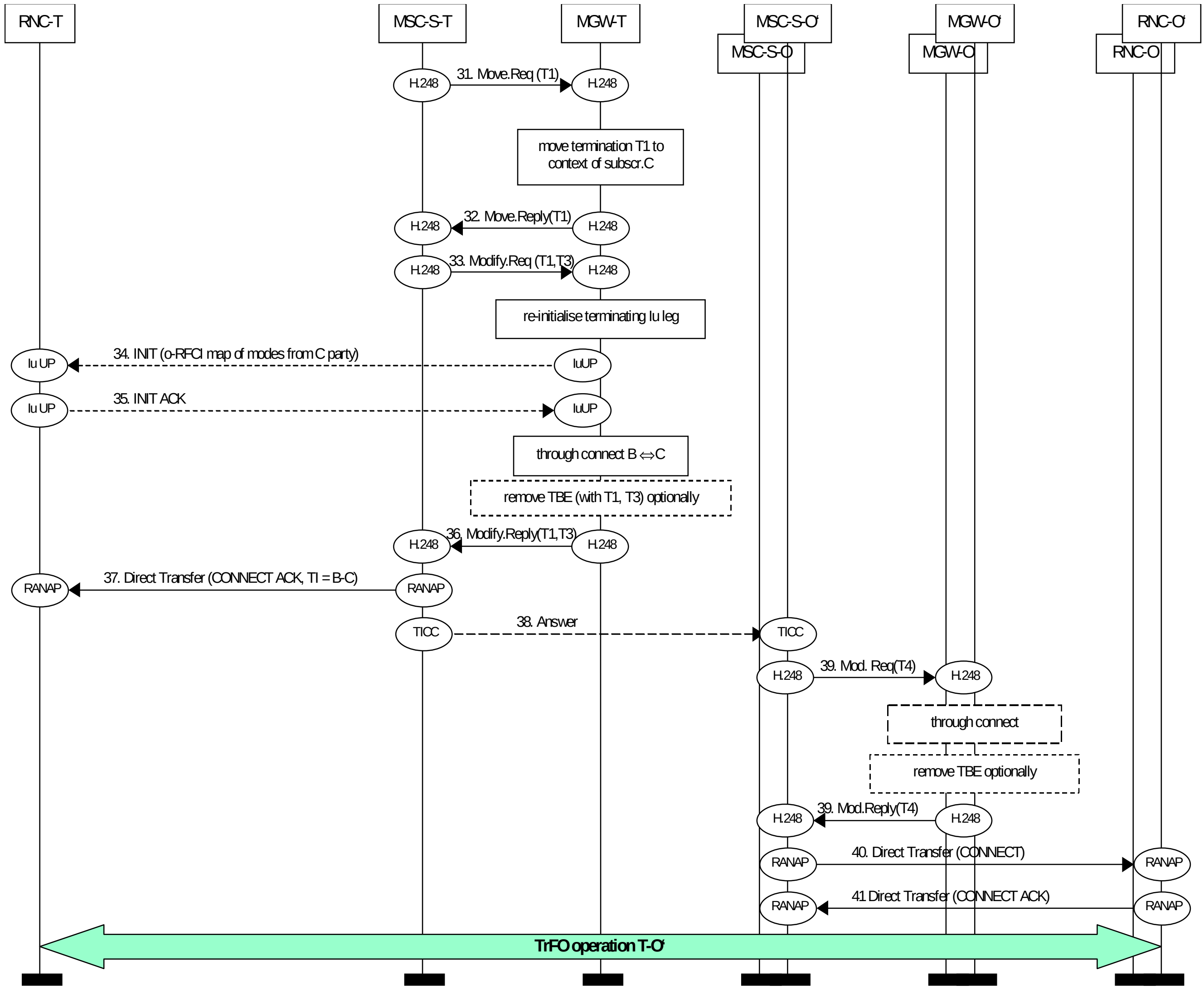


Figure 6.6/5: Call Hold/Call Wait and TrFO. Message flow part 4.

6.7 External Network to Mobile TrFO Call Establishment

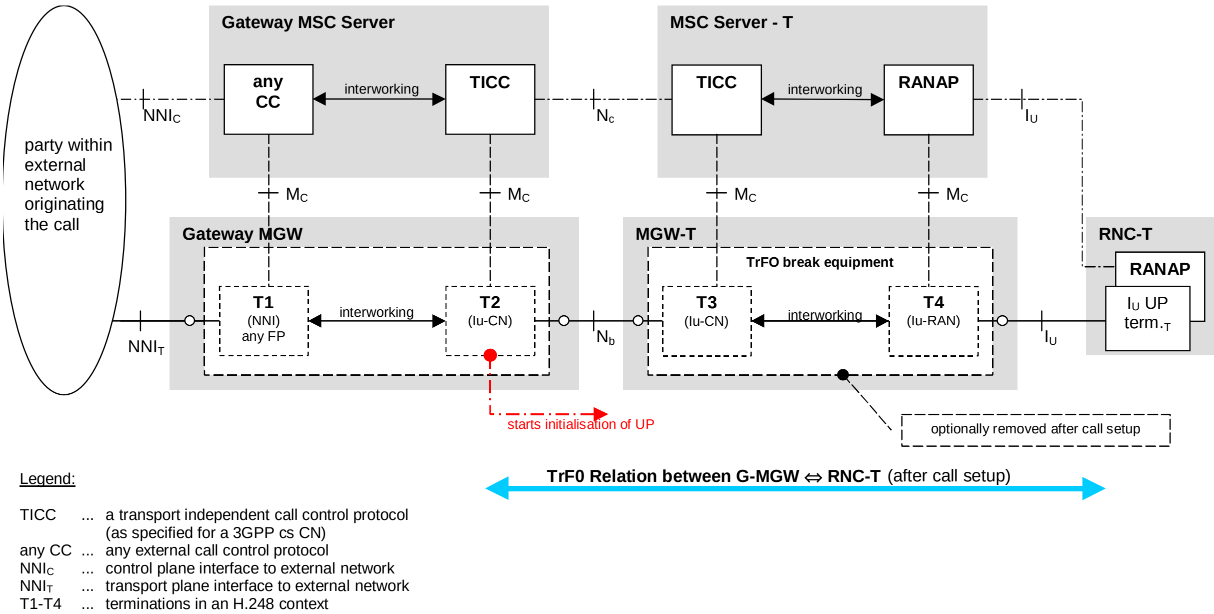


Figure 6.7/1. Configuration during Call Setup of a External Network to Mobile Call.

The description of Figure 6.1/1 (Configuration during Call Setup of a Mobile to Mobile Call) within chapter 6.1 applies for the network and protocol entities involved in the External Network to Mobile Call scenario with following modifications:

No RNC-O is present – a party served by an external network originates the call instead

The originating CN nodes are Gateway nodes (Gateway MSC Server / Gateway MGW)

The Gateway MGW call context is no TrFO break equipment in general, i.e. T1 in general do not support the IuUP framing protocol. Appropriate interworking (in some cases transcoding) has to be performed between T1 and T2.

Therefore Figures 6.1/2 to 6.1/4. (the respective message flows for mobile to mobile call setup) apply in principle as well with appropriate modifications outlined below:

Codec negotiation

Step 1. Until 6., that give the codec negotiation phase in Figure 6.1/2, shall be applied with following modifications:

There is no originating UE involved in this negotiation phase

If the preceding node of the Gateway MSC-Server doesn't support OoBTC procedures for compressed voice types, the Gateway MSC-Server shall initiate OoBTC procedures in order to enable transcoders placement at the edge gateway node.

The edge gateway node shall always send the complete list of the codec types and modes it supports for this type of call setup.

UP initialisation

The main difference compared to the Mobile to Mobile call setup is, that the CN side termination of the Gateway MGW (T2 in figure 6.7/1) shall start the initialisation of the IuUP according to the result of the codec negotiation. The forward initialisation principle shall be followed in any setup scenario.

6.8 Mobile to External Network TrFO Call Establishment

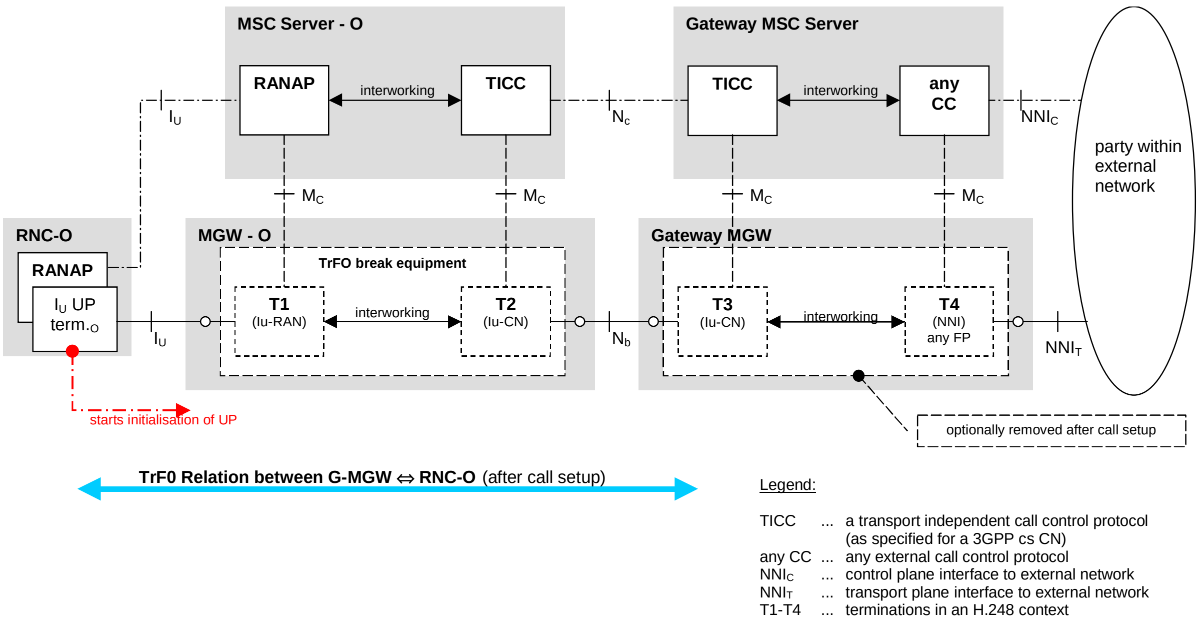


Figure 6.8/1. Configuration during Call Setup of a Mobile to External Network Call.

The description of Figure 6.1/1 (Configuration during Call Setup of a Mobile to Mobile Call) within chapter 6.1 applies for the network and protocol entities involved in the External Network to Mobile Call scenario with following modifications:

No RNC-T is present – a party served by an external network is the terminating side of the call instead

The terminating side CN nodes are Gateway nodes (Gateway MSC Server / Gateway MGW)

The Gateway MGW call context is no TrFO break equipment in general, i.e. T4 in general do not support the IuUP framing protocol. Appropriate interworking (in some cases transcoding) has to be performed between T3 and T4.

Therefore Figures 6.1/2 to 6.1/4. (the respective message flows for mobile to mobile call setup) apply in principle as well with appropriate modifications outlined below:

Codec negotiation

Step 1. Until 6., that give the codec negotiation phase in Figure 6.1/2, shall be applied with following modifications:

There is no terminating UE involved in this negotiation phase.

If the succeeding node of the Gateway MSC-Server doesn’t support OoBTC procedures for compressed voice types, the Gateway MSC-Server terminates the OoBTC procedures in order to enable transcoders placement at the edge gateway node.

The edge gateway node shall accept the codec MSC-O prefers and shall not puncture out any codec mode.

7 Interactions with supplementary services

7.1 Call Deflection service (GSM 03.72)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclause 6.3 is applied.

7.2 Line identification services (GSM 03.81)

7.2.1 Calling Line Identification Presentation (CLIP)

No impact.

7.2.2 Calling Line Identification Restriction (CLIR)

No impact.

7.2.3 Connected Line Identification Presentation (COLP)

No impact.

7.2.4 Connected Line Identification Restriction (COLR)

No impact.

7.3 Call forwarding services (GSM 03.82)

7.3.1 Call Forwarding Unconditional (CFU)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclause 6.3 is applied.

7.3.2 Call Forwarding on mobile subscriber Busy (CFB)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclause 6.3 is applied.

7.3.3 Call Forwarding on No Reply (CFNRy)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclauses 6.3 is applied.

7.3.4 Call Forwarding on mobile subscriber Not Reachable (CFNRc)

In order to apply the confirmation tone to the originating party, the speech insertion procedure described in subclauses 6.3 is applied.

7.4 Call wait (GSM 03.83)

In order to apply the notice tone to the interjected party, the speech insertion procedure described in subclause 6.4 is applied.

7.5 Call hold (GSM 03.83)

In order to apply the notice tone to the held party, the speech insertion procedure described in subclause 6.4 is applied.

7.6 Multiparty (GSM 03.84)

In order to mix calls, the speech insertion procedure described in subclause 6.4 is applied.

7.7 Closed user group (GSM 03.85)

No impact.

7.8 Advice of charge (GSM 03.86)

No impact.

7.9 User-to-user signalling (GSM 03.87)

No impact.

7.10 Call barring (GSM 03.88)

7.10.1 Barring of outgoing calls

No impact.

7.10.2 Barring of incoming calls

No impact.

7.11 Explicit Call Transfer (GSM 03.91)

In case that a call A-B is transferred to C by B (A-C as result), A-B may use codec x, A-C may use codec y, the procedure described in subclause 6.3 is applied.

7.12 Completion of Calls to Busy Subscriber (GSM 03.93)

No impact.

8 Charging

The selected codec shall be included in all the call data records of the call legs involved in out-band codec negotiation belonging to a particular subscriber.

Annex A (Informative): Status of Technical Specification 23.153

Status of Technical Specification 23.153		
Date	**Version**	**Comments**
September 1999	0.0.0	First draft prepared by the rapporteur
October 1999	0.1.0	2nd draft prepared by the rapporteur (Updated version from Abiko.)
November 1999	0.2.0	3rd draft prepared by the rapporteur
December 1999	1.0.0	Submitted to CN#06 for information
February 2000	1.1.0	4th draft prepared by the rapporteur
February 2000	1.2.0	5th draft prepared by the rapporteur (Updated version from Milan.)
February 2000	1.3.0	6th draft prepared by the rapporteur (Updated version from Milan.)
February 2000	1.4.0	7th draft prepared by the rapporteur (Updated version from Milan.)
February 2000	1.5.0	8th draft prepared by the rapporteur (Updated version from Milan.)
March 2000	2.0.0	Submitted to TSG CN#07 for approval
October 2000	2.0.3	9th draft prepared by the rapporteur (Updated version from Windsor)
November 2000	2.1.0	10th draft accepted, input to TrFO workshop #5, Stockholm
November 2000	2.1.1	11th draft, workshop #5 interim editors document.
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Text and figures: **Stylesheet:** 3gpp_70.dot **Rapporteur**: Toshiyuki Tamura / Phil Hodges **Company:** NEC Corporation / Ericsson L. M.		

Appendix A (informative): Codec Re-negotiation

A node may perform a procedure (e.g. handover) that results in a completely new list from that which was originally negotiated. Assuming that the current Selected Codec is still common (no Selected Codec Modification or re-negotiation) then the node shall send a Re-negotiation Request with the new Supported Codec List. The Supported codec list may then be punctured by nodes in the network in the same was as for the basic Codec Negotiation procedure and a new Available Codecs List returned.

If a node performs a procedure (e.g. handover) that results in both a completely new list and also the need for a new codec then Codec Re-negotiation may be performed with a request for a new codec selection. The procedure is then the same as for an initial codec negotiation.